DENIAL OF DEAF? DANGER IN THE STUDIO!

SEPT/OCT 1990 DISPLAY UNTIL 10/31

EQ

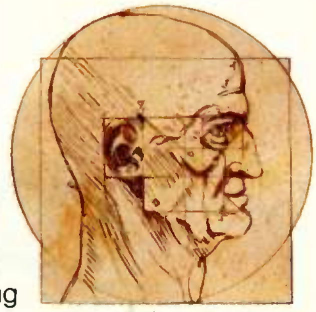
THE CREATIVE RECORDING MAGAZINE

MIKING
TIPS & TECHNIQUES

- **MIKING:** VOCALS, GUITAR, HORNS & DRUMS
- **PLUS!** A GUIDE TO 50 TOP MIKES
- **PRODUCER ROMA BARAN:** IN SYNC WITH 'STRANGE ANGEL' LAURIE ANDERSON
- **IN REVIEW:** EUPHONIX AUTOMATED MIXER COMP/LIMITER COMPARISON

$3.95
Can. $4.25
U.K. £2.50
A GPI PUBLICATION
10
0 262937 7

LEXICON THE ART AND SCIENCE OF SOUND



Leonardo Da Vinci was one of those rare individuals to whom both artistic and scientific excellence came easily. The concept of digitally reproducing and shaping analog sounds would have fascinated him.

Achieving excellence in digital audio technology requires a combination of scientific precision and artistic interpretation

that is as rare today as it was in Leonardo's time. It requires mastery of both the art and the science of sound.

Lexicon has devoted nearly two decades perfecting clean, natural digital audio sound without the quirks that digital technology is often accused of. It's called the **Lexicon Sound**, and you'll find it in every product we make.

When Lexicon introduced the world's first digital audio product in 1971, it was heralded as a major technological achievement, a breakthrough. We've built on that reputation with other achievements, like reverberation and time compression/expansion, that have earned us more than a dozen awards, including an Emmy.

Today Lexicon products serve more digital audio applications than ever before. There is OPUS, the world's only fully integrated random access digital audio production system. OPUS replaces a room full of equipment with a comfortable console and single equipment rack. Digital editing and mixing has never been smoother, more accurate, and more natural.

And everyone has a LARC. The Lexicon Alphanumeric Remote Console harnesses the power of the 480L Digital Effects System. The 480L produces a broad variety of superb effects from a growing library of software. It's no wonder you'll find LARC and the 480L in virtually every major recording studio throughout the world.

Lexicon's LXP-1 and LXP-5 effects processors provide a compact, economical way to include the Lexicon Sound in your productions. Each processor provides significant effects capability all by itself. But combined, the number and breadth of their effects is astonishing.

The CP-1 Digital Audio Environment Processor extends the Lexicon Sound into the living rooms of America. It's been lauded as one of the truly great digital audio products of all time, and

Stereophile ranked it as "one of a handful of genuine advances in sound reproduction".

These are just a few samples of Lexicon's digital expertise. Our full range of products serves the needs of recording engineers, musicians, film and video producers, radio and television broadcasters, as well as discerning audiophiles. In each product we've merged

technical excellence and artistic elegance. The result is the Lexicon Sound.

For more information about Lexicon products or a demonstration of their complete capabilities, contact us at (617) 891-6790, FAX (617) 891-0340, or write Lexicon, Inc., 100 Beaver St., Waltham, MA 02154.

lexicon

The Art and Science of Sound

EQ

VOLUME 1, ISSUE #4 SEPT/OCT 1990

THE CREATIVE RECORDING MAGAZINE

FEATURES

IN PROFILE: CRAIG DORY *By Freff*
Classical record producer divulges some secrets of live recording.....20

CRANK IT DOWN *By Cary Tennis*
They warned us that rock 'n' roll would make us deaf. Were they right? *EQ* takes a look inside your ears.....22

IN PROFILE: ROMA BARAN *By Ted Greenwald*
The big science of producing Laurie Anderson.....36

EQ WORKSHOP: MIKING TECHNIQUES *By Michael Marans*
Omni or cardioid, small or large diaphragm, condenser or dynamic? Top engineers share tips for miking vocals, acoustic guitar, horns, and drums.....51

51
COVER STORY

Maybe you use a locker full of microphones every day in the studio. Or maybe those old U47s are gathering dust in a back room. Either way, our studio microphone primer will hone your miking chops. *PLUS:* Special Applications Guide with application tips for over 50 microphones!

IN REVIEW

EUPHONIX CRESCENDO AUTOMATED MIXER *By Brent Hurtig*.....68

SHURE VP88 STEREO MICROPHONE *By Scott Martin Gershin*.....73

APHEX EXPRESSOR, ORBAN 412A, & JBL/UREI 7110 COMPRESSOR/LIMITERS *By Leslie Ann Jones*.....74

FOSTEX 2016 LINE MIXER *By Tom Ford*.....76

DEPARTMENTS

EDITOR'S NOTES.....4

REFLECTIONS.....8

LINE IN/LINE OUT How to reach *EQ*.....9

GUEST EDITORIAL Is there life after ghostwriting?.....10

UPDATE
Subliminal seduction, The Belgian Connection, new gear, and more.....14

HOW IT WORKS
By Bryan Lanser & Rick Eberly
The anatomy of a hard disk drive.....60

STUDIO REFERENCE SERIES
By Bryan Lanser & Jeff Burger
Digital format conversion chart.....65

ADVERTISERS INDEX.....78

CLASSIFIED ADVERTISING.....79

COLUMNS

FUTURE WATCH *By Linda Jacobson*.....8

THE SHARP ANGLE *By J.D. Sharp*.....12

ACROSS THE BOARD *Roger Nichols*.....80

CREATIVE WORKSHOP

IT'S ABOUT TIME *By Norman Weinberg*
Create rhythms with arpeggiators.....34

HANDS-ON PRODUCTION *By Michael Marans*
Beyond VCAs: Let MIDI be your automator.....35

DOWN TO BUSINESS

IN THE POCKET *By Steve Harlan*
More basics of studio business organization.....45

TECH SUPPORT

STUDIO CLINIC *By Richie Moore, PhD*
Fluxing your recording muscles.....47

VISUAL MEDIA

FILM & VIDEO SOUND *By Larry Blake*
Don't get mired in your effects library.....48

THE BIG PICTURE *By Robert Wait*
Visualize music for better scores.....63

PHOTO: FB STIMSON/PICTURED: STEPHANIE PASTERNAK WITH NEUMANN M249

Record A Track, Go To Jail?

THE COLD WAR IS OVER, AND NORIEGA IS IN THE SLAMMER. WITH its foreign enemies disappearing quickly, the American New Right is turning its agenda stateside to the arts and artists, such as the talented Robert Mapplethorpe and the questionably talented 2 Live Crew.

It *is* easy to be offended by the rap lyrics of 2 Live Crew. Songs about violence, forced sex, and male dominance of women are horrifying to me. (Of course, those same things can be found in most middle-of-the-road movies, which no doubt are viewed and enjoyed by some of the same judges and sheriffs who would censor 2 Live Crew.) But is censorship and court action the correct approach?

Right now, police are arresting musicians and the people who sell certain "obscene" records. How long before a sheriff shows up at some studio and arrests an engineer or producer for "aiding and abetting" the creation of obscene material? For that matter, how long before a studio is shut down, similar to the closing of a printing press that reproduces obscene material?

If you're a working engineer or producer, chances are you've participated in the creation of something that *someone, somewhere* could consider offensive. Are you willing to go to jail for your work? What sort of work would you turn down, for your own moral reasons, or to avoid going to jail? There aren't any easy answers.

Josef Skvorecky knows a thing or two about this issue. He is a Czech writer who endured ongoing censorship in his country until he emigrated to Canada in 1968. In his brilliant book, *The Bass Saxophone* (published in 1980 by Lester & Orpen Dennys of Toronto), he tells how Eastern European musicians were sent to Nazi death camps for "defiling musical culture." Skvorecky also recalls from that time a set of ten rules. Here's a few of them:

> "—So-called jazz compositions may contain at most 10% syncopation; the remainder must consist of natural legato movement devoid of the hysterical rhythmic reverses characteristic of the music of the barbarian races and conducive to dark instincts . . . (so-called 'riffs');
> —On no account will Negroid excesses in tempo (so-called 'hot jazz') or in solo performances (so-called 'breaks') be tolerated;
> —Preference is to be given to compositions in a major key and to lyrics expressing joy in life rather than Jewishly gloomy lyrics;
> —Musicians are likewise forbidden to make vocal improvisations (so-called 'scat')."

If you find those rules frightening, buy the book and post all ten rules next to your effects rack; it might not be long before they're replaced by new rules for the '90s. Next issue, *EQ* senior editor Elisa Welch Mulvaney will explore this controversy further.

ON A MUCH MORE CHEERFUL NOTE, *EQ* CONTINUES TO GROW. I'D LIKE TO introduce you to two people who have joined our staff:

Debbie Greenberg is our new editorial assistant. Formerly with *Music Technology*, she recently moved from Los Angeles to work with us here in the San Francisco Bay Area. Debbie's broadcasting background and journalistic skills are great assets to us.

Kristine Turnipseed is our new associate art director. Kristine comes to us with magazine design and production experience, and her designs already have found their way into *EQ* (gracing the Producer's Tips and Back Tracks in our Profiles).

We welcome both to *EQ*.

Brent Hurtig

EQ

EDITORIAL DIRECTOR
PHIL HOOD

EDITOR
BRENT HURTIG

MANAGING EDITOR
LINDA JACOBSON

ASSOCIATE EDITOR
JEFF BURGER

SENIOR EDITOR
ELISA WELCH MULVANEY

EDITORIAL ASSISTANT
DEBBIE GREENBERG

ADVERTISING DIRECTOR
DAVE SWEET

ASSISTANT ADVERTISING DIRECTOR
RANDY ALBERTS

CLASSIFIED ADVERTISING
AMY MILLER

ART DIRECTOR
SAROYAN HUMPHREY

ASSOCIATE ART DIRECTOR
KRISTINE TURNIPSEED

GRAPHICS ASSOCIATES
ELIZABETH LEDGERWOOD, RICK EBERLY

GROUP PUBLISHER
CHARLES C. BAAKE

BUSINESS MANAGER
PAT CAMERON

DIRECTOR OF PRODUCTION
ANDREW A. MICKUS

DIRECTOR OF CIRCULATION
JERRY M. OKABE

CIRCULATION MARKETING MANAGER
PERRY FOTOS

DIRECTOR, SINGLE COPY SALES
JAKE HUNTER

SINGLE COPY SALES ASSISTANT
CYNTHIA SMITH-ALLEN

GROUP ADVERTISING DIRECTOR
ED SENGSTACK

GROUP PROMOTIONS DIRECTOR
DAVID WILLIAMSON

ADVERTISING PRODUCTION COORDINATOR
PETER MRYZGIOCKI

ADVERTISING PROMOTIONS COORDINATOR
SUSAN ITO

ADVERTISING SALES ASSISTANT
ART THOMPSON

MILLER FREEMAN publications
A MEMBER OF THE UNITED NEWSPAPERS GROUP

CHAIRMAN OF THE BOARD, **CLIFFORD D. JAKES**
PRESIDENT, **MARSHALL W. FREEMAN**
SENIOR VICE PRESIDENT, **CHARLES C. BAAKE**
SENIOR VICE PRESIDENT, **LEONARD E. HAAS**
SENIOR VICE PRESIDENT, **THOMAS L. KEMP**
SENIOR VICE PRESIDENT, **WINI D. RAGUS**
VICE PRESIDENT, **CHARLES H. BENZ**
VICE PRESIDENT, **ANDREW A. MICKUS**
VICE PRESIDENT, **DONALD A. POZOUR**
VICE PRESIDENT, **LEIGH M. FREEMAN**
CHIEF FINANCIAL OFFICER, **STEPHEN A. DANIELS**
TREASURER, **MILLER FREEMAN III**
ASSISTANT TREASURER, **DOUGLAS M. DENNY**

EQ (USPS 002-952, Volume 1, Number 4; ISSN 1050-7868). EQ is published bimonthly by Miller Freeman Publications, 500 Howard Street, San Francisco, CA 94105. Telephone (415) 397-1881; FAX (415) 995-2486; Telex #278273. Please direct advertising and editorial inquiries to: EQ, 20085 Stevens Creek, Cupertino, CA 95014. Telephone (408) 446-1105; FAX (408) 446-1088; ITT Telex #4494425. Application to mail at second class postage is pending at San Francisco, CA. Subscription rates for the U.S. are $27.97 for 6 issues (1 year). All orders from outside the U.S. must be accompanied by payment in U.S. funds with additional postage of $5.00 for surface mail or $20.00 for air-mail. All subscription orders, inquiries, and address changes should be sent to P.O. Box 57295, Boulder, CO 80322-7295. For quickest service, telephone toll-free (800) 234-6072; in Colorado, (303) 447-9330. Please allow 6-8 weeks for address change to take effect. Postmaster: send address changes to EQ, P.O. Box 57295, Boulder, CO 80322-7295. EQ is a registered trademark of Miller Freeman Publications.

16 mm microfilm, 35 mm microfilm, 105 mm microfiche and article and issue photo copies are available from University Microfilms International, 300 N. Zeeb Rd., Ann Arbor, MI 48106. Telephone (313) 761-4700.

Audit Bureau of Circulations Member

Introducing the DigiTech Super Harmony Machine

The DigiTech IPS-33B Super Harmony Machine lets you create 1- and 2-note harmonies that are crystal pure and correct to scale. And that's just the beginning.

Check this list of super features:

- 1 and 2 note harmonies; above, below or either side of the note played
- 7 superb effects—harmonies, stereo pitch detune, up to 1.5 ms of digital delay, stereo chorus, up to 2 octaves up or down note and chord pitch shifting, volume modulation, pitch correction
- Harmonies smoothly follow note bending
- 24-bit processor ensures you can't outrun your harmonies
- User-programmable harmonies
- 80 different arpeggiations
- 128 presets, 128 user-definable programs
- Harmony shifting in real time
- 128 user-definable memory locations
- Chromatic harmonies in any interval up to two octaves up or down in any key
- Intelligent scallic harmonies in 59 different scale types, in any interval, in any key
- Choose chordal harmonies that fit chords
- Create special harmonies
- Make key and preset changes from the IPS-33B, the optional footswitch or your instrument
- Independent left and right effects mixing
- Adjust to MIDI keyboard chords
- MIDI continuous control of programs, patches and parameters
- 42.5 kHz sampling rate
- Tune instrument to IPS-33B or vice versa
- String synthesis
- Two continuous controller inputs
- 32 character backlit LCD display
- Remote hand controller included
- 20 Hz to 20 kHz bandwidth
- 90 dB signal-to-noise ratio
- Less than 0.03% THD

Rent the Super Harmony Machine demo video for $10.00 refundable deposit. Send deposit to the address shown below.

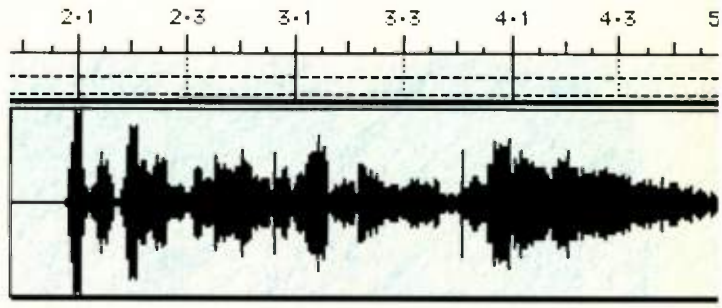
DigiTech
5639 South Riley Lane
Salt Lake City, Utah 84107

DigiTech®
The Sound of the 90's

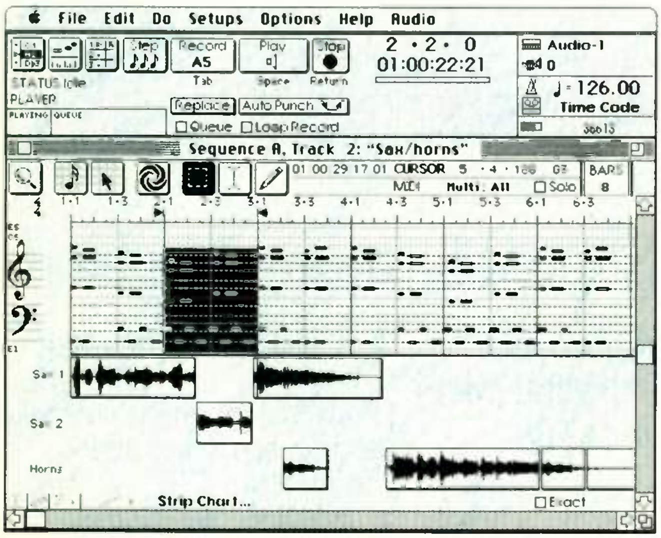
A Harman International Company

Opcode Now Gives

Introducing Studio Vision
Opcode's Vision sequencing program for the Macintosh was voted "Best Music Software Innovation of the Year" by the readers of Keyboard Magazine. Wait'll they check this out. Studio Vision combines all the features of Vision with the ability to record CD-quality audio direct to disk along with your MIDI data. Studio Vision works in conjunction with Digidesign's Sound Tools Digital Recording System and runs on any Macintosh II series computer or the SE/30.



Studio Vision has non-destructive editing of the 16bit digital

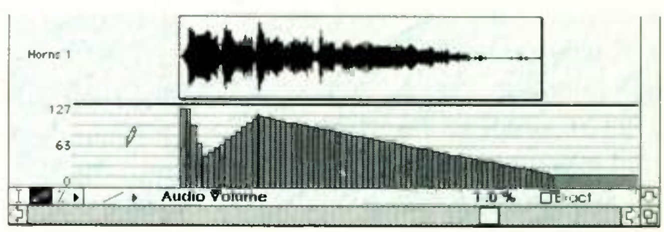


The control bar and graphic editing screen of Studio Vision.

Using Studio Vision and Sound Tools, you can play back two mono digital audio channels simultaneously—with the ability to record as much audio as your hard disk space allows. The program incorporates the speed and convenience of non-destructive editing of the digital audio right along with the MIDI—use commands such as cut, copy, paste, clear, merge, and strip silence. Studio Vision includes SMPTE synchronization and full automated mixing of the digital audio tracks with pan and volume control. Add Digidesign's DAT I/O digital interface for compatibility with most professional digital audio tape recorders. For a simpler setup, Studio Vision also works in conjunction with Digidesign's Audiomedia card. The power of Studio Vision with either hardware system eliminates most multitrack syncing situations when combining MIDI and "live" tracks.

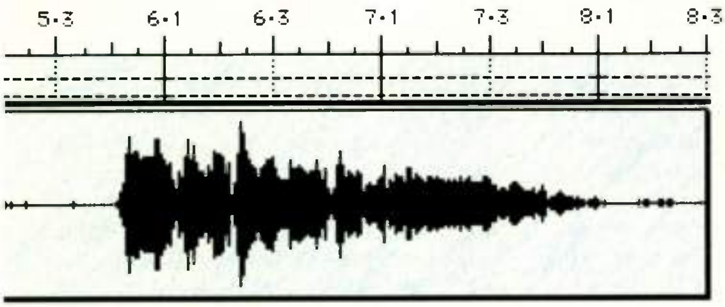
Record Vocals Too!
Studio Vision is the future of computer music recording. It's not just for MIDI keyboards, it's for any "live" instrument—guitar, saxophone, or vocals! Recording artist Thomas Dolby says: "You've got the best sequencer and the best recorder combined into one... you have an idea and ten seconds later you hear it... your creativity as an artist is not trampled on by technology slowing you down."

Use your standard MIDI instruments to record "basic tracks" like drums, bass, and keyboards. Then use the digital audio tracks to record the guitar, vocals or sax solo. Record a few takes of the sax solo and cut and paste from each of them to get the perfect take. Record one great chorus and paste it at each chorus in the song. Merge and offset background vocals for richness. Studio Vision's extraordinary flexibility allows you to record separate takes of digital audio onto any number of tracks.



Automated mixing of volume and pan in Studio Vision.

Voice to Your Vision



audio channels, plus all the editing commands in Vision!

Studio Vision dynamically allocates them onto the two playback channels. Now you've got your whole song on the Mac—from instruments to vocals.

And Studio Vision is ready for the next steps in technology: as the Macintosh and DSP (Digital Signal Processing) hardware get faster, Studio Vision will play back more than two channels at once.

Post Production

For post production, Studio Vision combines MIDI sequencing with the ability to record "voice overs" integrated with the music. Using SMPTE sync you can immediately check audio and video, and perform intricate audio edits instantaneously. Cut out a cough or noise here, swap sentences there, and adjust the music to fit the contour of the dialog all on the same screen. Isn't it nice to use just one computer keyboard?

Audition sound effects stored on hard disk before opening them, choose the one you want and place it on a specific SMPTE number, then lock up to video. Convenient.

List editing with SMPTE times of digital audio sound effects.

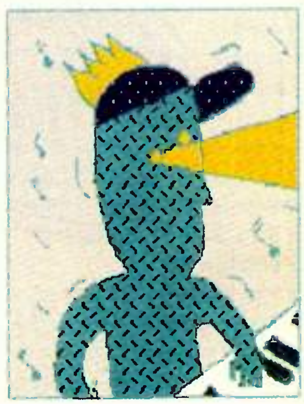
What Price Perfection?

You may be thinking this is awesome—but how much? With a Macintosh II (or SE/30), Opcode's Studio 3 (or any MIDI interface), Digidesign's Sound Tools or Audiomedia, and a hard disk, you're up and running with Studio Vision. And for an integrated MIDI setup, Studio Vision works closely with Galaxy, Opcode's universal librarian.

So if you've been waiting for that really big breakthrough in music technology, or if you're still using that same old MIDI sequencer synced to tape—think smart and go to your local Opcode dealer and hear how Opcode has brought voice to the Apple Macintosh with Studio Vision. Call Opcode for a free brochure and the name of your nearest dealer.

Studio Vision™

Specs



- *Full MIDI sequencing capability with graphic and list editing*
- *16bit 44.1Khz direct-to-disk recording (mono: 5 megabytes per minute)*
- *Playback of two digital audio channels*
- *Recording of digital audio limited only by hard disk space*
- *Non-destructive editing of digital audio*
- *Simultaneous integrated editing of MIDI and digital audio*
- *Full automated mixing of digital audio and MIDI tracks*
- *Compatible with all digital audio formats*
- *Digidesign Sound Tools or Audiomedia owners need only purchase Studio Vision*
- *SMPTE synchronization (except with the Audiomedia card)*
- *Upgradable from Vision*

Opcode systems, inc.

Opcode Systems, Inc.
3641 Haven, Suite A
Menlo Park, CA 94025-1010
(415) 369-8131

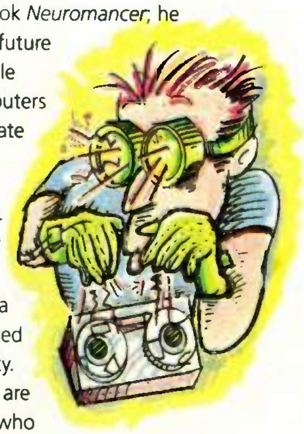
REFLECTIONS

future watch

Virtual Reality Check

By Linda Jacobson

CYBERARTS INTERNATIONAL (Sept. 6-9, Los Angeles) represents an effort by *EQ* and *Keyboard* to establish our own corporeally bound dimension of cyberspace. Cyberspace? The word was coined by author William Gibson in his sci-fi book *Neuromancer*; he wrote of a future when people using computers will interrelate in a "complex consensual hallucination," called cyberspace—a jointly created virtual reality.



If drugs are for people who can't handle reality, then virtual reality is for people who want to manipulate reality without handling drugs. VR pioneer Jaron Lanier resists the drug comparison—what if someone wants to regulate VR?—but the new technology does enable . . . well, consensual hallucinations. We now can enter that plane of existence thanks to artificial intelligence, robotics, animation, and the people who love computers.

To experience VR, you don dark, bulky goggles and a wire-covered glove, becoming one with a computer. The goggles contain two tiny video monitors with a synthetic, 3D view of an environment—a living room, perhaps, or underwater tableau—rendered in low-res color graphics. The software-generated image is carried via cable, which also sends data about your head move- ▸▸

I read Ira Cord Rubnitz's guest editorial in the July/August issue of *EQ*, and I must say I disagree with most of what he says. I am not a digital fanatic, nor an analogophobe. I like each for its strengths, and have different preferences depending on the situation. Yet I found Mr. Rubnitz's problems with editing, copying, and assembling digital product perplexing. I have not had these problems myself, nor have I commonly heard about them. I have found digital open-reel razor blade editing, particularly on the Sony 3202/3402, to actually be easier than analog editing, in that the acceptable error width—the amount by which you can be "off" on either side of your edit point—is much greater. It's kind of like editing analog at 90 ips.

Those engineers who are most successful with digital have not tried to transfer too much knowledge from analog to digital, but have been willing to humble themselves, to experiment and relearn, and to *unlearn*, so that they can deal with digital on its own terms. Just ask Neil Dorfsman, or *EQ*'s own Roger Nichols, whose digital recordings and knowledge are arrived at, I am sure, through hard work and conscientiousness. The problems Mr. Rubnitz is having may be very real, but they are not digital's fault.

DANIEL LEVITIN
San Anselmo, CA

Regarding the "Tangled Story Of DAT" [*EQ* May/June] by Cary Tennis, it's amazing that an article of this length could so thoroughly avoid actually explaining the different sides of the DAT/copyright issues. If I wanted to know the "distance between Nags Head and Washington," I'd get out a map and figure out the mileage. If I wanted to try to make up my mind about the DAT/copyright issue, I would hope to read material that really explains the different sides. This article does not. It should have been placed under "guest editorial," not "cover story."

My opinion of the rest of the magazine, including Brent Hurtig's sidebar on copy codes, remains the same—I thoroughly enjoy it.

DAVE HANNER
Pittsburgh, PA

I thoroughly enjoyed your May/June issue with the exception of one item which "stuck in my craw." The sidebar "15 Arrangements That Broke The Rules" ["Arranging For The Studio"] was an informative and handy reference. I just got irritated at the John Lennon quote, "If George Martin's such a genius, why don't people buy *his* albums?" Such a statement assumes that the degree of mass appeal determines whether a musician is a genius or not. This is like saying that because McDonalds sells billions of hamburgers, the food is of master-chef quality.

L.C. BEBBI
Baltimore, MD

Thanks for being the first magazine I've seen with the guts to put together a comparative review of near-field monitors [*EQ* Mar/Apr '90]. This type of review is one of the most useful services a magazine can provide for its readers. I get little out of separate reviews that end with nice round words such as, "If you're looking for a nice-sounding speaker at a reasonable price, you should seriously consider speaker XYZ."

You will probably receive a number of criticisms from readers who feel that no source can be adequately qualified to pass comparative judgments on audio gear. Like it or not, *every* piece of audio gear is somewhat subjective, and I believe using a panel of ten audio engineers in a double-blind test will yield results that are less biased and less subjective than the alternative.

BARRY MANDEL
Rochester, NY

I read with interest "The Electric Near-Field Acid Test" in your March/April issue. I spend most of my life in front of the little beasties and don't have the time or money to buy every speaker you reviewed. On the basis of your admittedly subjective evaluation, I acquired, against my better judgment, a pair of Peavey [AMR] PRM-308s. I mean, just the logo, you know?

Boy, was I wrong. As you say, the speakers have to be heard to be believed. So far I've mixed two albums through them (Steve Earle and Colin James, both released in June) and neither I nor the artists could be happier with the results. Thanks for the tip and, if you're in the market for several pairs of NS-10s, please give me a call.

JOE HARDY
Ardent Recordings
Memphis, TN

Our customers have expressed numerous complaints regarding your comparison of studio monitors [*EQ* Mar/Apr]. We are also confused by your choice of evaluation procedures as they relate to actual mixing. Your subjective listening results came out predictably in that the brighter and more bass-pronounced models did well. But is it the duty of the monitor to give a pleasant loudness curve, or provide an accurate reference point from which a mix can be built?

This leads us to [ask], what is a studio monitor?. . . In previous models we used different reflex alignments, brighter tweeters, higher system

▸▸ PAGE 78

ILLUSTRATION: SCOT HALPIN

LINE IN/LINE OUT

THIS ISSUE'S REVIEW OF THREE compressor/limiters was authored by Leslie Ann Jones. The daughter of bandleader Spike Jones, she is a 15-year veteran engineer and mixing specialist. This year she served as one of the house mixers at the Grammys. Her most recent project involves Dave Edmunds, on Capitol Records, where she is a staff engineer.

Roma Baran's profile is the work of Ted Greenwald. Ted is a recording artist and producer in New York, as well as a former technical editor of our sister publication *Keyboard*. Look for Ted's byline in an upcoming *EQ* article about the convergence of MIDI and digital audio. •

How To Reach EQ

Writing To *EQ*

We welcome comments, opposing views, praise, criticism, slings, arrows, and even spurious aspersions—all receive an equal hearing at *EQ*. We reserve the right to edit letters for space and clarity, and please, if you don't want your letter published, tell us so in writing. All letters become the property of *EQ*. Write to Reflections, *EQ*, at our Cupertino address, listed below.

Writing For *EQ*

If you'd like to write something for us, send a query letter with samples of your writing and information about yourself to editor Brent Hurtig at *EQ*. No calls, please. We rarely purchase unsolicited manuscripts, so write to us before sending your story. We try to read all submissions within ten days. Please allow up to three weeks for a reply.

Tips & Solutions

EQ is always looking for unusual ideas for our Studio Reference Series, How It Works, and Studio Solutions departments. Please send your ideas (you'll be rewarded if we use them) to managing editor Linda Jacobson at *EQ*.

Technical Assistance

If you're having trouble with a piece of equipment or a studio procedure, we can't offer telephone assistance (have you called the manufacturer?), but we *will* provide answers to some of your questions in upcoming issues of *EQ*. If you have a recording question or problem, send it to: Question Authority c/o *EQ*.

Complaints About Advertisers

If you bought a product that is advertised in *EQ* and are dissatisfied with it, but cannot resolve your problem, write (don't call) editorial director Phil Hood at *EQ*. Include copies of all relevant correspondence and please be as specific as possible in providing details of the problem.

New Products & Reviews

Send new product information to be included in a particular issue to "Update," c/o Jeff Burger at our Cupertino office, at least three months prior to the cover date. We also review recording and production equipment, both hardware and software. Send record/video/book release information to "Reviews" c/o Linda Jacobson.

Permission & Reprint Rights

To request permission to use all or part of an article appearing in *EQ*, write to Peggi Clapham at *EQ*. For advertising reprint rights, send your request to advertising director David Sweet at *EQ*.

Advertising

All display advertising is handled by advertising director David Sweet and assistant advertising director Randy Alberts. Classified advertising is handled by Amy Miller. They can be reached at our Cupertino office. (Telephone (408) 446-1105; Fax (408) 446-1088. Office hours, 9-5:30 PST.)

Overseas Advertising

Our advertising representatives for the British Isles and Scandinavia are David Landsdowne and Melissa Thomas. Write or call Landsdowne Media Services, Ltd., Landsdowne House, 12 Church Street, Rickmansworth, Herts. U.K., WD3 1BS. Telephone (internat'l.) 44 923 777711/(nat'l.) 0923 777711. Telex 28764 LMSLTD G. Fax (internat'l.) 44 923 721103/(nat'l.) 0923 721103.

Contact us at our main address: *EQ*, 20085 Stevens Creek, Cupertino, CA 95014.

EDITORIAL DIRECTOR
PHIL HOOD

EDITOR
BRENT HURTIG

MANAGING EDITOR
LINDA JACOBSON

ASSOCIATE EDITOR
JEFF BURGER

SENIOR EDITOR
ELISA WELCH MULVANEY

EDITORIAL ASSISTANT
DEBBIE GREENBERG

◂◂ ments back to the computer. Swing your head and you see things above, below, behind, and before you. The sensor-lined glove, cabled to the computer, lets you move around the room, grab and manipulate objects, defy gravity, and interact with other virtual visitors.

VR first manifested itself in 1981 as "Super Cockpit," a flight simulator on an Air Force base. Former Air Forcer Thomas Furness ushered in VR for the military to engage in kill-the-virtual-enemy games. He recently enthused to *Discover* mag that with VR, we can change the world. However, he warned, VR can lead to "the creation of socially immature people. Virtual realities will do what people want them to do, and that's not the way the real world works."

That contrasts with the thinking of Jaron Lanier. This young entrepreneur, along with cohorts at his Silicon Valley firm VPL Research, invented the aforementioned EyePhone ($9,400) and DataGlove ($8,800), among other VR hardware and software components. Lanier envisions VR as a tool for self-transformation. At VPL Research, you can go anywhere and be anyone or anything; you can trade perceptions with a friend, or live as a lobster—complete with claws. Lanier says, "Virtual reality is shared and objectively real like the physical world, composable like a work of art, and unlimited and harmless like a dream. When VR becomes widely available around the turn of the century, it will not be seen as a medium used within physical reality, but rather as an additional reality."

Architects, educators, scientific visualizers, space explorers, war-game simulators, flight trainers, and surgeons now are, or soon will be, reaping the benefits of that reality—although it still is far from being perfected. VR nonetheless presents endless opportunities for arts and entertainment.

When VR was conceived in '65 as computer-created imagery allowing 3D vision and movement with ▸▸

The Ghost In The Music Machine

This guest editorial originally was signed by its author. After we decided to publish it, the author was advised by lawyers to withdraw it, due to contractual obligations made by the author when hired as a ghostwriter. Intrigued by the situation, we decided to publish the piece and guarantee its author complete anonymity.

What causes a person to enter the music business? Usually, an intense love of music. Unfortunately the second word in the phrase "music business" sometimes interferes with that love, causing many musicians to question their original intentions.

The music business is especially tough on composers. To obtain work, a composer must have experience in the field. Experience is usually backed by credits—film scores, jingles, TV themes, and the like. The big question is, how can a composer gain work without previous credits? One way to acquire composing experience is by ghostwriting.

When composers get too busy to handle their workload, they occasionally hire an outside composer—a ghostwriter—to help complete a particularly heavy schedule. This outside help means anything from writing the last two bars of a 32-bar theme, rewriting a cue for a TV show, arranging and embellishing a piano piece for a rock band, to composing an entire score.

Would You Give Up The Ability To Claim You Had Worked On A Recording Project?

Even though it is common practice for composers to use ghostwriters, most people I talk to are naive about its use in the industry. I am not saying that every well-known composer uses outside help to complete his or her work; it's just that there are more than a few composers who do this.

As a ghostwriter for various music houses and composers, I've been wrestling with the issue of authors' rights. A standard ghostwriter's agreement basically strips a composer of any rights to the music he or she ghostwrites. Sometimes these agreements even prevent ghostwriters from claiming they had worked for the composer or music house in question. Understandably, the word "ghostwriter" implies that the ghosting composer should remain invisible, but how can a fledgling writer gain credits, and advance in the industry, if prevented from claiming credible experience that is rightfully his or her own?

Is there a good side to ghostwriting? For a novice composer with little or no industry background, ghosting can offer lessons in writing music, and how to deal with the business side of composition. All the formal music training in the world cannot take the place of real-world experience.

What about the composer who has plenty of experience in the field, but not many credits to show for it? This is where ghostwriting becomes extremely frustrating. Being a musician and composer, I attach a bit of ego to just about anything I create, so it is difficult for me to know that my music is being sold under someone else's name. Before starting a ghostwriting relationship with someone, I try to determine what will be fair for both parties, in terms of monetary compensation and claims to the music involved.

An agreement between composer and ghostwriter must be stated at the beginning of a project. Ask these questions:

• If there are publishing residuals, are you entitled to any? Sometimes a composer

▸▸ PAGE 66

The author is a composer and engineer who ghostwrites for several television and film score composers.

ILLUSTRATION: REYNOLDS T. SHOMAKER

If you've been putting off doing stereo field remotes for fear of risking a fragile, expensive stereo mic, Shure's new VP88 is what you've been waiting for.

The VP88 is an advanced single point stereo condenser mic that not only recreates the sonic environment with extraordinary audio fidelity, but meets Shure's legendary standards for ruggedness and reliability.

The VP88 is built to withstand the punishment of field remotes. And, it comes at a price you'll find surprisingly affordable.

TRUE MS STEREO.

The VP88 features a forward facing Mid capsule, perpendicular Side capsule and built-in stereo matrix to assure a wide, natural, uncolored response for stereo imaging. Yet, it's perfectly mono compatible.

To enable you to control the degree of stereo spread and ambience pick-up, the VP88 has three switch-selectable stereo modes or direct mid and side output. And it's designed to provide the wide dynamic range and low noise you need for remote broadcasts.

THE FEATURES YOU NEED.

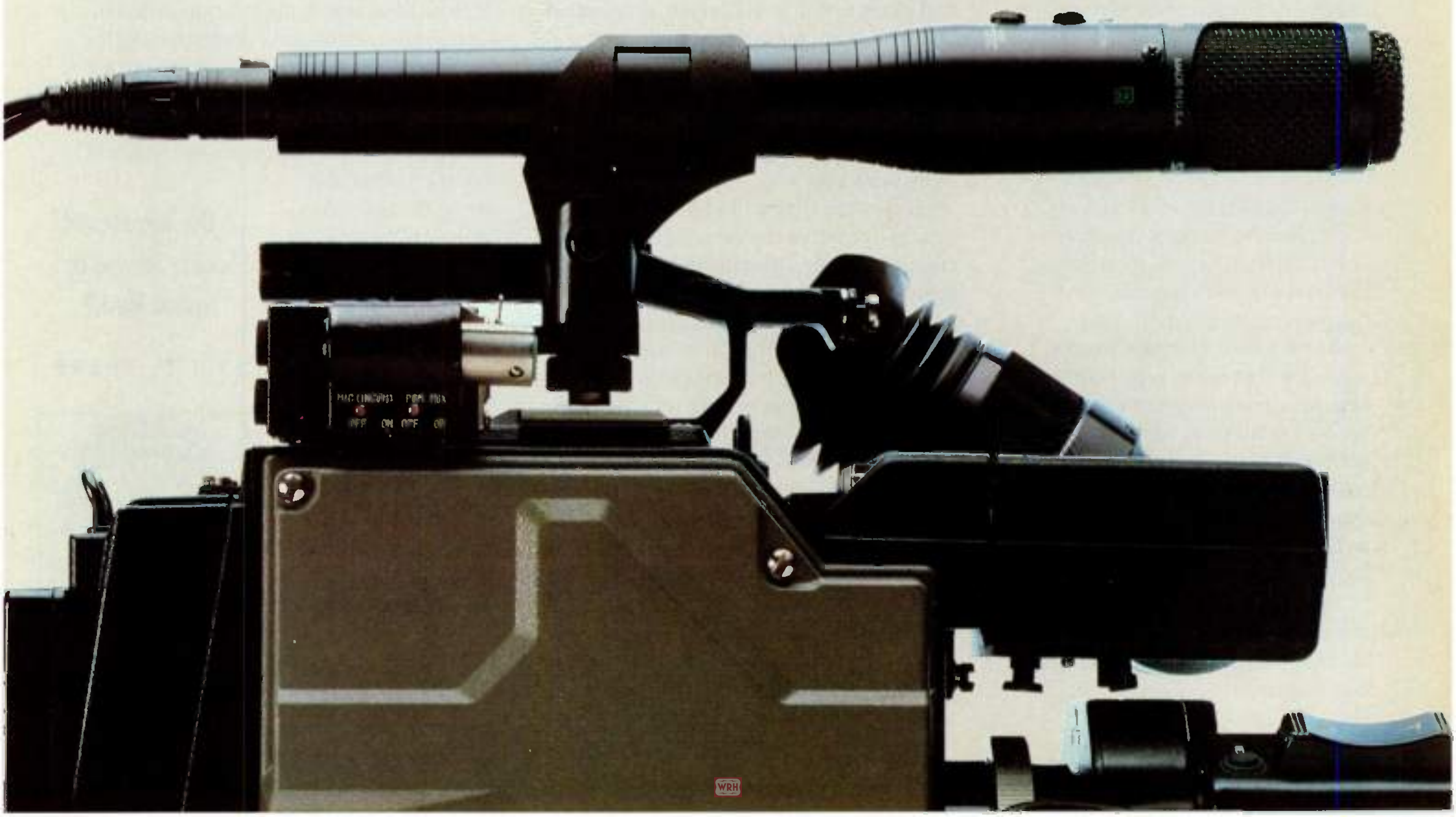
The VP88 can be powered by a self-contained battery or phantom power so you can go where the action takes you. It includes switchable low-frequency rolloff for reduced ambient noise and a built-in "pop" screen.

In addition to camera mounting, the VP88 can be used on a stand, fishpole, or boom. And the mic comes with a wide range of standard and optional accessories to accommodate your most challenging stereo miking requirements.

So whether you're just beginning to look at stereo miking, or you want to take your stereo to the next level — consider the advantages of the Shure VP88. It's making stereo miking an affordable proposition.

For the name of your nearest dealer and our free brochure, call or write Shure, 222 Hartrey Avenue, Evanston, IL 60202-3696. 1-800-257-4873. The Sound of the Professionals®... Worldwide. SHURE®

Shure's New VP88 Stereo Microphone Offers A New Level Of Reliability And Affordability.



◄◄ tactile sensation, no one mentioned anything about *hearing*. Then came Lanier, a musician who collects instruments and has composed TV soundtracks. One of VPL's first projects used a "Z-Glove" to literally and audibly play air guitar. Now Lanier imagines EyePhones incorporating tiny headphones: "The sounds are a bit unusual in that they're processed to have 3-dimensional quality; they come from certain directions." Meanwhile, folks at NASA have found a way to use the DataGlove to play a virtual drum kit. Peter Gabriel, Brian Eno, Laurie Anderson, and other musicians are exploring VR to create new forms of art.

What about VR and recording? Maybe one day you'll put on your MixerGloves to create a virtual mix. You'll hear the results of signal processing and fader moves without committing them to reality, much less tape.

Better yet: Imagine producing and engineering a live recording session composed of artists who are physically located in different cities, but gather, via VR technology, in one virtual "studio" to cut tracks. No Ted Koppel's *Nightline*-type action where I see you but you can't see me; virtual sessions will enable the eye contact and body language that contribute to music's overall groove and feel.

Back to today. At CyberArts, Jaron Lanier and his goggle-&-glove fashionware will be on hand, as will such virtual explorers as Autodesk (inventors of 3D drafting software), which plans to market a relatively low-cost Cyberspace system by 1991. Folks from Mattel will bring interactive toys; maybe we'll get to play with the $90 Nintendo game controller, the Power Glove—the first consumer VR product. (Like NASA's toys, it is based directly on VPL Research technology.) CyberArts also will showcase the virtual reality of 3D sound, and various computer-generated realities.

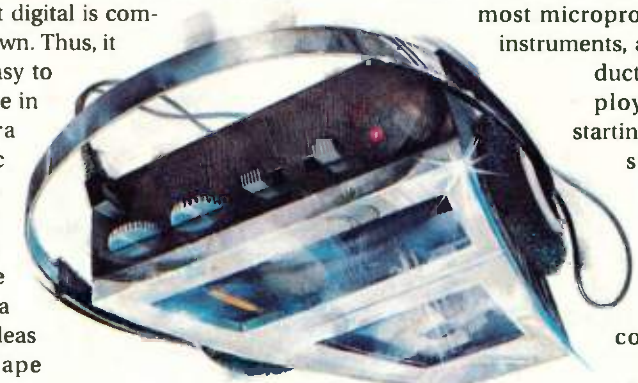
Maybe one day we'll all have access to the ultimate do-it-yourself kit for those who want to create their own reality. •

Does Technology Kill Creativity?

Is Our Microwaved Society Serving Up Instant Music?

BY J.D. SHARP

THE TECHNOLOGY FOR the production and recording of music is ever-improving, as is the price/performance ratio of that technology. Not a month goes by, it seems, without some new high-end innovation once again raising the standards, while new relatively low-cost products bring such capabilities as 24-track recording and 36x24 mixing into a new range of affordability. The digital recording era has dawned at perhaps a more languid pace than some might have guessed, but digital is coming into its own. Thus, it would be easy to assume we're in a golden era of music production.



Except for one thing. There seems to be a paucity of ideas worth the tape (or disk) on which they're recorded. Somehow, as the equipment improves by leaps and bounds, the material captured on it becomes more and more trivial, at an equally accelerated rate. This is not meant to denigrate some of the world-class efforts issued periodically; it is just that this era in popular music is far from being golden, and could well be one of the most forgettable periods in pop history. The question is, why?

The cause appears to be twofold. First, technology, rather than serving the needs of creative artists, is dominating them. Much in the way that many Hollywood movies consist of a collection of chase scenes and special effects, edited in a rapid-cut TV "soundbite" style, pop songs increasingly are an ear-catching collage of audio gimmicks and "hot" drum samples with little or no lyrical and melodic content to please the intellect and touch the soul. In a way, the manufacturers are to blame for making their equipment *too* good. In synthesizer days of yore, when names like Moog and ARP dominated the scene, a player had to know a great deal just to get the instrument to play in tune, much less actually sound like something other humans would want to hear. Now it's difficult to hear something bad come out of a synthesizer, as increasing numbers of keyboardists are spoon-fed patches by a handful of top programmers who define the synthesizer sound-texture fashion statement of the month. And few modern drum machines sound anything but great.

Never mind that the result has been that most microprocessor-based instruments, and most productions that employ them, are starting to sound the same. As long as tapes and CDs keep selling, why worry? The reason to be concerned is that the gimmicks will get old, and the public will tire and move on to other forms of entertainment.

In an ideal world, the solution would be a renaissance in the art of programming. It's encouraging to note the production of new synthesizers such as Yamaha's SY77 and Roland's D70, instruments deep enough to provide powerful platforms for original sound creation. But these things take time, and in a microwaved society there may be no minutes left to explore the intricacies of sonic texture.

The second factor contributing to the flatness of popular music is a loss of musicianship. Our time-constrained lives, and our impatience from decades of TV-watching and instant gratifi-

►► PAGE 66

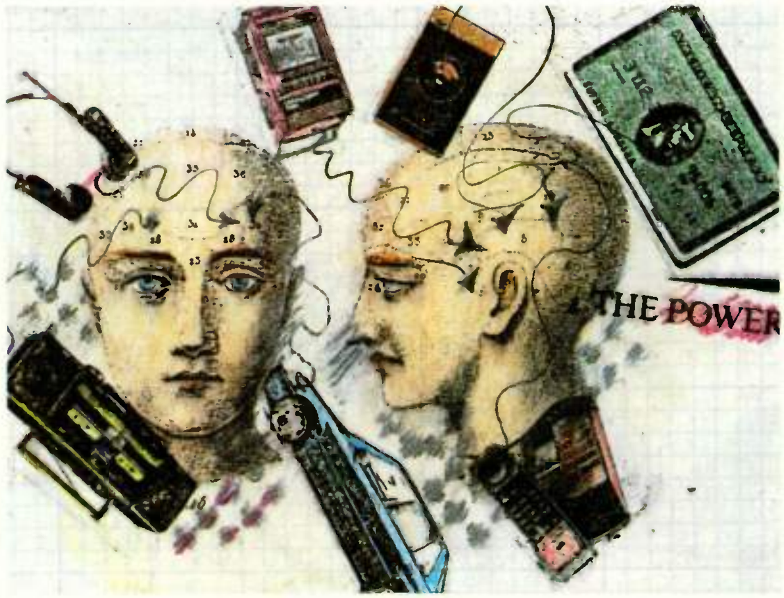
J. D. Sharp is the owner and proprietor of Bananas At Large, a Northern California pro audio and music dealership.

ILLUSTRATION: DAVE MARRS

UP DATE

News & Notes From The World Of Modern Recording And Knob Twisting

Compiled by *EQ* editors



Music That Eats Your Brain?

A DECADE AGO, A controversy raged over subliminal messages in visual advertising. While subliminal programming in audio has received less attention, it most definitely is big business: Self-help messages are masked behind soothing nature sounds and "purchase-encouragement" messages are embedded in music for playback in stores. Up until now, this programming has taken a back seat in dB to foreground material.

Enter Silent Subliminal, a new trademarked process developed by **Lowery Associates**, which allegedly transmits messages at inaudible high frequencies in a format that our ears readily can decode. While the listener doesn't perceive any sound consciously, affirmations are claimed to hit the eardrums at a *minimum* of *115dB higher* than standard masked levels. You can bet they're strongly perceived by some of our animal friends.

Somehow, encoded tapes can be played on standard cassette decks, but according to the inventor, the signal cannot be duplicated using standard tape-copying techniques.

While Orwellian images leap to mind—of corporate and political propaganda zorking our brains—the Silent Subliminal developers describe several applications for music, beyond the obvious "I-really-like-this-music" messages. It is said that enhanced dimensionality can be gained through techniques such as duplicating the musical programming on the subliminal tracks with a slight delay, or using subliminal tracks for echo and ambience.

If you're worried that you'll start walking out of stores with products you don't want, rest easy: Silent Subliminal signals probably could be jammed. Check out that guy in the next aisle with the personal satellite dish strapped to his head . . . .•

Hello, AES!

THE **89th Audio Engineering Society Convention** takes place September 21-25 at the Los Angeles Convention Center. Of the many major presentations scheduled for the gathering, several stand out as relevant to *EQ* readers: Technologist Keith Johnson will discuss the various areas in which he has contributed significantly, including magnetic recording, control systems, loudspeaker design, high-speed cassette duplication, and motion picture sound; Jim MacDonald—the voice of Mickey Mouse for over 50 years—will share his experiences in designing thousands of sound effects for movies and cartoons; and a student workshop will bring together representatives of several audio recording schools.

Panel discussions abound, as usual. *EQ* columnist Larry Blake will chair a forum on the use and future of DAT in productions. Top sound designers will participate in a "round table" conversation about sound for today's small movie theaters. Disney's Terry Porter will describe techniques for processing and cleaning up dialog. Master engineer/producer George

ILLUSTRATION: SAROYAN HUMPHREY

Gems By The Thames

THE APRS—THE U.K.'S "Professional Recording Association"—holds a trade show each year in London; this last June *EQ* made its debut at APRS. Close to 200 exhibitors happily packed into the inconveniently located, miserably non-air-conditioned Olympia Hall. Here's a rundown of some highlights:

Tannoy introduced six new British-built speakers. The Monitor Series ($550/pr. to $5,500/pr.) boasts many new features, including new designs for coaxial drivers, cabinets, and internal wiring. The Canadian-built PBM Series will continue to be available in North America. Outboard processing specialists **Drawmer** were also on hand. Of note is their new DL241 Auto-Comp compressor/limiter/expander/gate ($649), designed to compete with the Rane DC24 and other cost-effective dual-channel dynamic controllers. Full manual control is available, although an AUTO switch lets the unit run with little adjustment required. There's no SIDE-CHAIN ACCESS, but hey, it's a Drawmer. In other rack-sized products, **Motionworks** displayed the Midiworker ($TBA), which brings MIDI event control to SSL consoles.

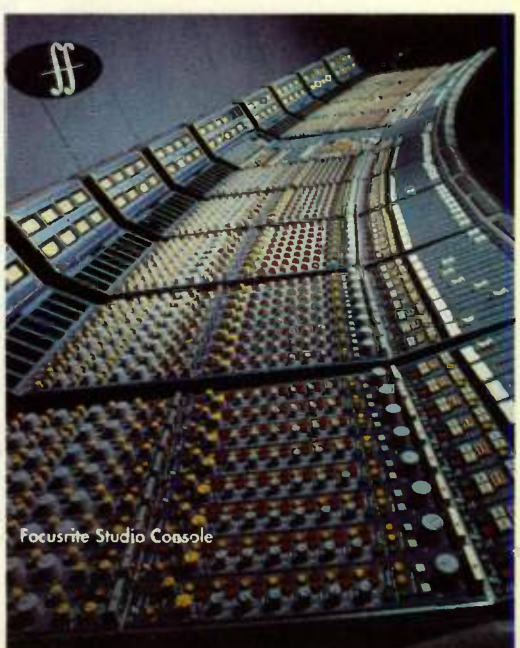
Big-ticket equalizers, compressors, and more were on hand from **Focusrite**, **Solid State Logic**, **GML**, **Neve**, and others, including a new outboard contender—**Amek**. Industry legend Rupert Neve now enjoys *carte blanche* design privileges at Amek. The first of his rack-mount products is the 2-channel Medici equalizer ($6,018), which features both WARMTH and SHEEN controls (we heard 'em, they work). In conjunction with a MICE box ($1,900), Medici offers full MIDI switching control. Six grand is a lot for an EQ. But manufacturers are looking down-market from "world class" facilities to the enormous number of working 24- and 16-track facilities that could benefit from the addition of high-end processing, either on a rental or purchase basis.

Speaking of Mr. Neve, Amek's Mozart console (see Update, May/June '90 *EQ*) now is available with Neve-designed EQ sections. The Mozart was the first mixer to offer an "all-input" (tape-monitorless) option, recognizing that many people monitor tape tracks through full input channels. Two other "all-input" boards of note made their APRS debut: **Focusrite** kicked off the show in British style with a champagne toast (at 11:00 AM!) around their massive Studio Console ($TBA). The board sports 12 aux sends, 48 groups, and the well-known ISA110 EQ modules. It lacks any sort of session computer, but the presence of digitally assigned groups makes us wonder what's down the road. The **DDA** DMR12 ($35,500 to $41,500) is another all-input board (its 12 group output/tape return modules are identical to the input modules, but offer BUS TRIM instead of a mike preamp). MIDI muting and VCA grouping/automation are options.

AMS gave a most impressive demonstration of their Logic 1 console ($TBA), a fully automated digital mixer. As we suspected when we reviewed the 4-group Delta 200 console (May/June '90 *EQ*), **Soundcraft** has unveiled an 8-group version of their Delta Series. The Delta 8 ($TBA) is suitable for 8- and 16-track recording. **Studiomaster** has developed the fully modular, 24- or 32-input, 12-group Trackmix 24 console ($9,750 to $11,095).

And in the We-Can-Hardly-Wait-To-See-It-Working department: **Thompson Audio Developments** unveiled a mock-up of their T24 2" analog 24-track (around $20,000). This pinch-rollerless transport will be controlled by an outboard microcomputer. VU meters, track sheets, and even electronic calibrations all show up on the computer screen. The company mascot is a rubber duck. We hope it floats. •

Tannoy 215DMT



Focusrite Studio Console

Massenburg will spend an entire day sharing the secrets of his mixing techniques.

Additional scheduled seminars point to the hot trends in audio these days, covering the topics of computer music systems, studio management, loudspeaker assessment, sampling in post-production, the evolution of digital audio workstations, digital audio testing standards, acoustic design trends in listening environments, and interactive media training.

If you plan to attend AES, drop by the *EQ* booth for your free hearing test. The evaluation, administered by Bruel & Kjær, enables you to walk away with a print-out of your ears' frequency response. *EQ* wants to help you protect the most valuable tools of your trade. •

Where Neve Meets SSL

IN CULTURE, LANguage and politics, Brussels is a crossroads. It's a place where people pepper their sentences with both Flemish and French, where Cordon Bleu meets hearty peasant cooking, and where the nations of NATO convene. Perhaps it makes sense, then, for the Belgian capital to be home to the world's only SSL-equipped Neve console.

A narrow and old residential street is where Kitsch Studios can be found—a natural-lit, world-class studio environment run by Bruno Stevens and Theirry Van Roy that specializes in commercials and music projects. Two studios share a common machine room equipped with a Studer A820 24-track analog recorder and two Mitsubishi X-850 32-track digital recorders. Studio B has an older 32-input SSL Series E board; Studio A, designed by Andy Munro, has a 60-input Neve VR-Series board complete with George Massenburg Labs' moving fader automation, and remarkably, a Solid State Logic Series E Studio Computer—normally found only in SSL's own consoles.

It's at first a jarring sight to find the SSL Studio Computer—which most SSL users use to keep track of session information and for tape transport autolocation—sitting in the middle of a Neve. Why did Kitsch go to the trouble of installing this computer into a board from SSL's key rival?

"We had two SSLs, but needed to replace one since we needed more inputs," explains Bruno. "We were keeping one SSL, and wanted something else. We looked at the Amek APC1000, which is a great console, but there aren't many installed. What we really wanted was an SSL with moving faders and the Neve sound. So Neve built overdub switches above each monitor section for us, just like on an SSL. The GML system can read fader levels and mutes from an SSL floppy disk—so when it comes to cues and cuts and fader moves, this board is SSL-compatible. Add to that the SSL Studio Computer for session notes and autolocation, and you have the best of both worlds."

The Belgian Connection: Thierry and Bruno at Kitsch. Note the Neve's SSL computer and monitor display.

Adds Theirry: "Of course, our Neve doesn't have SSL Total Recall, to automate track routing and display knob settings. But we do have GML recall software [which is not compatible with SSL's recall at this time], and it's brilliant! Until we received it we used our own recall system for the Neve: a Polaroid camera!"

Was it difficult to integrate the SSL computer? "I'm sorry to disappoint you," laughs Bruno. "It takes about an afternoon to install, and in fact, there are no electrical connections between the SSL computer and the Neve. The hardest part was getting the cutout on the Neve panel. We bought the computer and its power supply from a London-based audio supplier and maintenance firm for about $16,000. Everything went very smoothly. I expect to see other studios follow our lead."

Kitsch calls themselves a "studio with atmosphere." That's for sure: One live room is filled with pews, gargoyles, and a pulpit, all from an ancient church. There's a kindly resident ghost, whom certain clients regard as a Muse. And the name of the studio? "When we started as a 16-track personal studio in 1982, we didn't have a lot of money, so we decided to decorate with kitsch items from the trash," says Bruno. "Plates with the Virgin Mary, ugly posters—at some point it was starting to severely affect the acoustics. Now we have cleared most of that stuff out, but there still remains an ugly plaster Belgian garden troll, so that we may remember our roots." •

Analog: The Format That Wouldn't Die

DOLBY LABORATORIES has introduced **S-type** noise reduction, designed for licensing primarily to makers of analog cassette machines. A Dolby spokesperson says, "When incorporated in a high-quality machine using today's best tape formulations, Dolby S-type provides analog cassette performance subjectively equivalent to digital media under home listening conditions." Also suggested was that S-type technology will find its way into 1" 24-track and 1/2" 16-track machines.

Like other Dolby technology, S-type involves encoding of signals during recording, and decoding during playback. S-type additionally yields reasonable playback quality—free from dynamic artifacts such as "pumping"—using Dolby B-type or no decoding.

S-type is based on the SR (Spectral Recording) process in use since 1986 on high-end professional multitracks at the cost of about $2,000 per channel. SR arguably is the main reason for the continuing viability of professional analog recording. S-type consists of a simpler three-chip (IC) set, and thus is available for consumer cassette decks at a cost (to the consumer) of $200-300. (Our sources report that there's a one-chip version in the works at Sony, which would cost considerably less.)

Both S-type and SR employ fixed and sliding frequency bands. Due to the cassette's concentration of noise in the high frequencies, and its relatively low print-through, S-type uses a single fixed band of low-frequency noise reduction. In the higher frequencies, two staggered stages of fixed and sliding band NR are used. (By way of contrast, SR uses two staggered stages of fixed and sliding bands on the low end, and three on the high

PHOTO: B. HURTIG

FOR THE RECORD

From our March/April issue: **Digital Designs** tells us that the version of their LS261s we included in our *Electric Near-Field Listening Test* has been out of production since November 1988. The current LS261s have been completely redesigned. Following on the heels of yet more improvements, the company is offering a woofer upgrade for owners of older LS261s. Contact Digital Designs at (619) 353-1290 for details.

From our July/August issue: **Alpha Audio's** Exabyte 8200 is a *hard drive*, the times associated with track-minutes apply to *stereo* recording time, and 8mm backup is about *140%* of real time. . . . The correct address for **AMS** is Billington Road, Burnley, Lancaster, BB11 5ES, England with sole U.S. distribution by AMS at 1108 Holm Road, Suite C, Petaluma, CA 94954; AudioFile pricing starts at $95,000. . . . A credit inadvertently was omitted for **Bruce Bartlett** as co-author of last issue's *Basic Tracks* column entitled *Mike Those Drums*. •

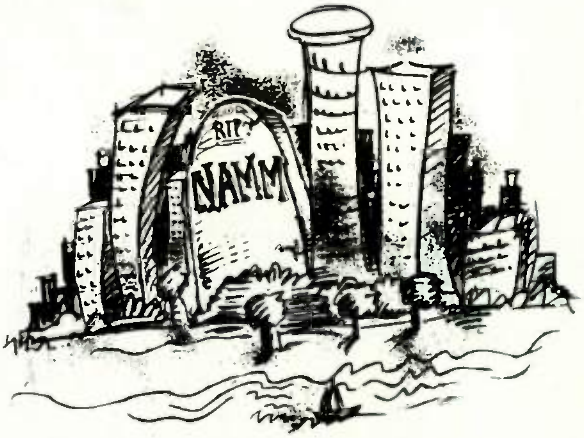
end.) The S-type technique results in up to 10dB of noise reduction in the low frequency range and 24dB for high frequencies.

Dolby Laboratories has published a list of requirements with which manufacturers must comply in order to license the S-type process, thereby insuring standardization and widespread consumer acceptance. These conditions include widened frequency response (50Hz to 14kHz), stringent azimuth-adjustment standards, tighter head-height calibration, and bias/record calibration adjustments.

The good news is that Dolby S-type noise reduction will elevate the sonic quality of cassettes to an all-time high. The bad news is that most consumers probably aren't looking for an excuse to buy yet another new high-end cassette deck—not when digital media is making the future of the audio cassette appear questionable.

In related news, **N.V. Philips**—developer of the cassette and CD format standards—has designed a new machine that records digital information onto standard audio cassettes. While Philips officials won't say much about it, the machine's sound quality reportedly is much better than analog, while not quite as good as CD. Its apparent advantage over DAT decks would be its ability to use the standard, low-cost cassettes familiar to us all. Also, manufacturers could sell similarly improved cassette decks at comparatively low prices by using the same transport they've been refining for decades. Philips' machine also can play standard cassettes in the analog domain.

Philips' progress in this area could be hindered by plummeting DAT prices, as well as such potential legal roadblocks as the copyright issues that delayed DAT's appearance in the U.S. marketplace. We'll keep you posted.•



Wake By The Lake

SNORE ON THE SHORE. GRAVE BY THE Waves. These were but a few of the nicknames that attendees hung around the neck of this year's Summer NAMM (National Association Of Music Merchants) International Music & Sound Expo in Chicago. The last few summer shows have been ghosts of their former selves, as more and more manufacturers have decided that one NAMM show a year was enough. The problem? It can cost some of the larger manufacturers upwards of a quarter-million dollars to truck in all their gear, transport and house the myriad factory reps, and wine and dine big clients—not to mention the disruption a show causes in the normal flow of business. After a few major companies began boycotting the summer show, the effect turned into a snowball with plenty of momentum. This year you could almost see the Grim Reaper looming over McCormick Place.

Not to worry: The Winter NAMM International Music Market in Anaheim, California, is alive and kickin' every January. And next year the summer show moves to New York City and will be expanded to include a special day for consumers to attend. Get ready for Exhilaration by the Station—Grand Central, that is. •

For addresses of companies mentioned in Update, please turn to page 78

ILLUSTRATION: SCOT HALPIN

like an amplified rim shot, since continuous exposure would come from various sites and sources. The fact that this program will be difficult to administer does not mean that it won't happen in North America, as well. Weidmann says, "Our industry had better monitor itself. It's getting way out of hand."

Concert promoters certainly need to be concerned. Sound pressure levels in the front rows of major rock concerts probably exceed the regulations set by the Occupational Safety & Health Administration (OSHA). Last year in New York City, a Supreme Court ruling gave local government the authority to control the volume level at concerts (although it was to keep neighborhoods from complaining about noise, not to protect anyone's hearing). Will this become a national trend? Will we soon start seeing health warnings on concert tickets?

We could not find any case of OSHA intervening on behalf of beleaguered studio workers. It may only be a matter of time.

BECAUSE OF THE DANGER OF GOVERNMENT INtervention, HEAR emphasizes education, not regulation. Its members fear that if the music and recording industries don't start policing their own ranks, some government agency eventually will step in. Just what we need: The Ear Cops.

Technical progress is dangerous. That is the lesson of our century. Whenever human beings sacrifice safety for profits, other human beings get hurt. Another lesson of our century: People who get hurt, sue. And some petition their government for a redress of grievances. We in the U.S. live in the most litigious and regulated society on the planet.

So it will happen. Some young engineer who learned to mix at the threshold of pain will lose his hearing and be out of a job. He's going to blame the company that hired him and the engineers from whom he learned his craft. His lawyer is going to find experts to say, yes, audio engineers all know what happens when you listen too loudly. And no, they did nothing to stop this young engineer from listening at ear-splitting levels night after night. And yes, a jury will decide the studio owes this guy something. They owe him, maybe, ten million dollars.

That's not going to be a good thing for music. It certainly will wake up anybody who hasn't gotten the message: High-decibel listening is aural suicide.

The pro audio industry is not unaware of this. Steps to protect hearing are being taken by engineers who walk out of the control room rather than listen at dangerous levels, by musicians who wear earplugs during rehearsal, and by doctors, acoustic engineers, and others who are putting the message in a more public form that can be understood by those who need to hear it.

The movements in Europe and the U.S. are encouraging. Let's hope the recording industry continues to raise its consciousness about ear protection. It would sure beat dealing with the Ear Cops and Audio Lawyers—or worse yet, risking the loss of the most valuable tools of our trade. •

IF THE RECORDING AND MUSIC INDUSTRIES DON'T START POLICING THEIR OWN RANKS, SOME GOVERNMENT AGENCY EVENTUALLY WILL STEP IN.

IT'S ABOUT TIME

The Return Of The Arpeggiators

BY NORM WEINBERG

ARPEGGIATORS—WHETHER THEY'RE designed into instruments or as stand-alone devices—can't be found as often as they once were. There was a time, in the late '70s and early '80s, when a musician merely had to hold down a chord, and let the arpeggiator step through the notes at a pre-determined pace. Now we tend to let sequencers do the trick, with step programming. Systems such as E-mu's Emulator and Emax still incorporate this feature. Let's talk about using arpeggiators like these to create percussion patterns. We'll use the Emax as a frame of reference, but you can apply the techniques here to many other products with arpeggiators.

FIG. 1

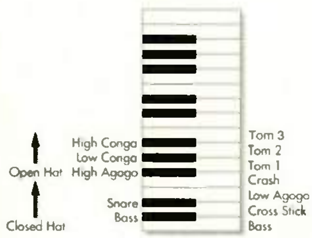


FIG. 2

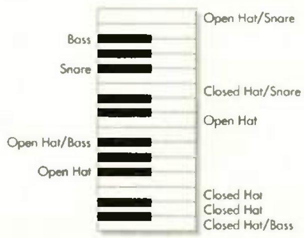


FIG. 3

The Emax arpeggiator has several programmable parameters, including tempo, note value, the arpeggiated interval, number of extensions, two additional harmony intervals, arpeggiation mode (up, down, up/down, forward assign, backward assign, and random), internal/external clock, and CRUZ control. Some of these features can be applied in most creative ways.

Many people think of arpeggiators in terms of melodic or harmonic functions. Let's say you program the arpeggiator to perform three extensions using the interval of a perfect fourth. If you hold down a *C*, the arpeggiator chooses from the pitches of *C*, *F*, *B♭*, and *E♭*. The exact ordering of pitches depends on the mode setting, while the rhythm is determined by the tempo and note value. If you held down both *C* and *E♭*, the notes of *E♭*, *A♭*, *D♭*, and *F* (an octave higher than the first *F*) are added to the collection of possible notes.

> Arpeggiators Can Help Create Innovative Percussion Tracks.

While arpeggios represent idiomatic techniques for instruments such as piano, guitar, harp, banjo, and mandolin, they can serve as creative partners if you're looking for some new beats or fill patterns for your drum tracks.

Method #1. Fig. 1 shows 12 drum sounds placed within the first octave. Four acoustic tom samples are placed under the notes of *C*, *D#*, *F#*, and *A*. Electronic tom sounds are placed under *C#*, *E*, *G#*, and *A#*, and the remaining notes have bass drum, snare drum, open hi-hats, and closed hi-hats.

When the arpeggiator is programmed to perform three extensions with the interval of a minor third, holding down the *C* will fire acoustic toms, the *C#* will fire electronic toms, and the *D* will create beat patterns. Experiment with the mode settings and the note values. For increased enjoyment, add a harmony setting of a major second. Now playing the *C* will call up acoustic toms along with the bass, snare, and hi-hats.

Method #2. The Emax can layer two samples on each key, enabling the arrangement shown in Fig. 2. In this case, closed hi-hats are placed under the first six notes of the octave, and open hi-hats are placed under the second six. Additional drum and percussion sounds are organized throughout the octave and layered on top of the hi-hat sounds.

This time we use an interval of a major second. This allows for five extensions within the octave, and two different percussion "kits." As shown in the diagram, playing the note *C* fires a bass drum, cross-stick snare, agogos, and congas. The *C#* produces sounds of a bass drum, snare, cymbal crash, and three toms. All the sounds are combined with a hi-hat (open or closed).

If you prefer ethnic patterns, try assigning several African or Latin American instruments to the keyboard. Realistic African grooves can be created by using a wide variety of shakers, talking drums, log drums, and mbira samples. Latin grooves can be created with timbales, congas, surdo, guiro, or cuíca samples.

Method #3. As you might expect, layering samples under each key can create a mighty busy texture. Most drummers omit a note or two, or play a hi-hat without the obligatory bass drum stroke. The trick to creating more interesting arpeggiated percussion patterns lies in leaving room for *silence*.

One way to create silence is to release the key. Fig. 3 shows another technique. We've placed a series of drum samples at intervals of a major third, taking care to leave some of the note locations empty. If we set the mode to random arpeggiation and ask for six extensions, the sampler selects randomly between firing five percussion

▸▸ PAGE 67

Norm Weinberg is an associate professor at Del Mar College in Corpus Christi, Texas. He has written extensively about the topics of electronic percussion and music education.

DIAGRAMS: RICK EBERLY

HANDS-ON PRODUCTION

Enhancing Your Automation Options

BY MICHAEL MARANS

ALTHOUGH YOU MAY BE MIXING ON A console with moving fader or VCA automation, sometimes it's more efficient to let MIDI handle the automation tasks for your sequenced tracks. With MIDI controller commands you can automate the volume, pan position, mute status, and, to some degree, the equalization of your virtual track instruments. All this can be accomplished with minimal fuss, and with unlimited freedom to edit and update your mix. In some cases, MIDI can do more than traditional automation, and do it faster.

Virtually all current, professional MIDI sequencing software packages can record MIDI controller data and system-exclusive messages. Typically, you enter the data either numerically from the computer keyboard, through real-time performance via a MIDI controller (like a footpedal or mod wheel), or graphically draw it with a mouse or with the sequencer's on-board MIDI faders in non-real time. Once the data is recorded into a sequencer, it can be edited in various ways, including compression/expansion and inversion. You also can copy and paste the data to any location on any track: Once you've perfected a mix move—a fade, for example—you can use the same data for all the tracks you wish to affect. (Remember to assign a new MIDI channel to the copy so it affects the appropriate track.) This can save a lot of time, and helps insure consistency from track to track.

MIDI Controllers Go Where No VCA Or Moving Fader Has Gone Before.

Volume Control. The MIDI spec provides for any controller number to be used for any function, but controller 7 generally is dedicated to volume. Most master keyboards let you assign the controller numbers that their sliders, wheels, and footpedals will generate, letting you manipulate the volume via the physical controller with which you're most comfortable. I prefer a data slider, as it most closely emulates the action of a fader. Some synths do not have fully assignable controllers, but in a pinch you can record the data as another controller number, then edit it to number 7. (Sequencers don't map controller data in real time, so you won't hear volume changes as they're recorded; you'll have to fine-tune them in edit mode.)

An important advantage of MIDI volume control is that volume changes can jump instantly from one *exact* level to another—a feat that's virtually impossible for a standard fader. Also, exact levels (127 in all) can be programmed, and changes can be locked to specific SMPTE time-code locations. You could, for example, fade in a track halfway through the chorus, instantly jump to maximum volume on the third beat of the fourth bar, then drop to an almost inaudible level on the third 32nd-note triplet after beat two in bar seven, then fade out from . . . you get the idea. Just enter the values manually at the desired points in the track.

Volume also can be controlled via the manipulation of note-on velocities. This is especially effective on tracks where the instrument's dynamic range is too great or too limited. In these cases, simply use velocity compression or expansion; it's a whole lot easier than trying to ride the track with a MIDI fader.

Panning. Controller number 10 is used for pan functions. The procedure for entering pan data is identical to that for MIDI volume control, though not all instruments respond to this command.

If your synth doesn't respond to MIDI control of its pan functions, all is not lost. Most synth pan functions can be modulated within the instrument via an LFO or MIDI velocity. In these cases, simply route the LFO or velocity to pan internally, then use MIDI controller 1 to control the LFO depth, or edit the note-on velocity data to achieve the desired effect. If you're modulating panning via velocity, be sure to decrease the sound's velocity-to-amplitude sensitivity, or you may find your pan moves adversely affect the dynamics of the sound.

Equalization. More and more synthesizers' sound parameters can be manipulated via system-exclusive (SysEx) commands. You can use these commands to control filter cutoff, operator output levels (in FM synthesizers), and on-board effects devices, among other things. To automate moves that affect EQ, you need only record the appropriate SysEx data. You can use a filter cutoff point—or the output level of an FM modulator—to make a sound brighter (or less bright). Many synths incorporate EQ in their on-board effects; you can use SysEx commands to tweak these tone controls. Once you record this data into a sequencer, you can edit it.

A Moving Experience. Unlike console faders, data-entry sliders typically let you record MIDI controller data just one track at a time. Several products have multiple sliders that can be mapped to appropriate MIDI channels, such as Lexicon's MRC (four channels), JL Cooper's FaderMaster (eight channels), and Blue Sky Logic's MIXI (16 channels). For an added twist, Niche offers a unit that controls the level of eight channels of standard audio using MIDI controllers!

It Ain't Perfect. . . . By using a lot of MIDI automation, you may overload the data stream, resulting in timing errors. Avoid this by using an interface with multiple assignable output ports to distribute data for different instruments over several lines, minimizing the error potential. You also can use your sequencer's controller thinning utilities to filter out all but the essential data. •

Sound designer and recording engineer Michael Marans is assistant editor of Keyboard *magazine.*

ROMA BARAN

IN AN INDUSTRY DOMINATED by hard-edged flash, Roma Baran stands apart as an approachable, exceptionally unassuming personality. Although she has produced some of the most exciting music of the past decade, her name isn't exactly on everyone's lips. Yet she has carved a distinct niche that is broad enough to encompass the avant-gardism of Laurie Anderson's work *and* the mainstream as represented by TV's *Oprah Prime Time*, for which she produced the title music. ■ Born in Poland amidst the upheaval of post-war Eastern Europe, Baran escaped to Canada at age 5 with her parents. As a teenager she played Ventures-style guitar in Montreal bands, eventually recording as a guitarist, keyboardist, and cellist with acclaimed singer/songwriters Kate and Anna McGarrigle. Baran then attended San Francisco Conservatory of Music, earned a Masters in music from University of New Hampshire, and completed Ph.D. course work in musical iconography (visual representations of musicians and instruments) at New York University. Her debut as a producer arrived when the trio Huxtable, Christiansen & Hood—fans of the McGarrigles—asked her to produce their record *Wallflowers* (on Philo Records) in 1977. "I had never produced or engineered before," Baran recalls. "But I'm in the habit of saying 'yes' to things." The result of this affirmative attitude was dubbed Record Of The Year by *Stereo Review*. ■ Baran then took a staff producer/engineer position at ZBS Media in upstate New York, the recording headquarters for Manhattan's avant-garde multimedia community, whose residents included Philip Glass, Meredith Monk, and Steve Reich. She achieved national prominence in 1979 with the unlikely success of Laurie Anderson's mesmerizing single "O Superman" from the Warner Bros. release, *Big Science*. The duo went on to assemble the post-modernist amalgam of styles, sonorities, and players that made their follow-up release, *Mr. Heartbreak* (Warner Bros.), the most successful meeting of pop music and New York avant-garde since Andy Warhol discovered the Velvet Underground. ■ Anderson's records established Baran as a producer of creativity and integrity, and also deepened her appreciation for the power of multitracking. "Working with Laurie rarely involves any predictable multitrack order," she explains. "She's likely to lay down some little element most people would think of as a late overdub, and make that the basis of the track. It freaked me out when I first started working with her, but over the years it's completely freed my sense of what could happen in what order." ➤➤

RENAISSANCE WOMAN IN AN AVANT-GARDE WORLD BY TED GREENWALD

PHOTO: REX MILLER

BACK tracks

Huxtable, Christiansen & Hood, *Wallflowers*, 1977

"At that time Philo Records, which was in an old barn in the middle of nowhere [in Vermont], had this accountant who was eventually fired because he kept doing really strange things. Nobody realized it, but for months he had been opening the window in the studio, putting bird seed on the sill, then closing it again. Birds would fly up to the window and scream and fight, and more and more records had birds on them. Eventually it turned into something of a disaster. They had to put up spikes and poison on the windowsills. We did *Wallflowers* in the middle of the disaster stage, so there are bird calls all over it."

Laurie Anderson, *Big Science*, 1982

"George Lewis came over to do a trombone part on 'Let X=X.' After he listened to it, he said, 'I've got a great idea!' He wrote out three incredibly complicated parts, which we recorded one at a time. Separately, they were just *bleep-bloop*, so we couldn't tell what the hell he was doing. I couldn't figure out how he was even reading it, because there were eighth-note rests in front of some of the downbeats, and sixteenth rests in front of others.

"When we were done, it turned out he had been counting 'one' on the offbeat! That's why it was so syncopated. He was hearing 'one' an eighth-note off. A lot of musicians have that problem with Laurie—even me. I played keyboards in one of the *United States* shows. After the show, Laurie said, 'That solo was very nice, but it seems to me that you were a little early.' I thought she was talking about some kind of rhythmic subtlety; maybe I was anticipating a bit, or something. But she meant *real* early, an eighth-note early. Over the years I've learned how she hears her music, how a normal musician would hear it, and how to discuss it so both of them will know what I'm talking about. Somebody once told me that the best way to count Laurie's music is, 'one—one—one—one . . . .'"

Various Artists, *Christmas Guitars*, 1989

"The concept for *Christmas Guitars* was ➤

Baran also devotes much professional energy to soundtracks, and has contributed to 17 film and TV productions, including music supervision and production for Columbia Pictures' Madonna vehicle, *The Bloodhounds Of Broadway*. But her most recent work involves the production of three records: Laurie Anderson's *Strange Angels* (Warner Bros.), which she took over in the form of "a big, expensive demo" after several producers had contributed to it with varying degrees of success; *Strange Cargo* (Private Music), a collection of high-tech tone poems by percussionist David Van Tieghem; and *Christmas Guitars* (Green Linnet), a release benefitting the National Coalition for the Homeless. This Yuletide package brings together fretboard talents as diverse as Michelle Shocked, Nile

"What irritates me the most is to be typecast by the way Laurie's records sound."

Rodgers, Al Kooper, Terry Roche, Larry Coryell, and cuatro virtuoso Yomo Toro, among others, in a charming, panstylistic collection of solo showcases.

As the antithesis of the *auteur* producer, Roma Baran tends to value an artist's idiosyncrasies over her own tastes. "You get the sense from some producers that they would rather be doing the vocals themselves. I don't have a style, or a *thing*, or a song aching to come out. That's not my interest in the process at all." Baran also says she is not interested in "art" music, despite her artsy, high-tech image. "I listen to country and pop. I like strong, clear-identity vocalists, like Roland Gift and Aaron Neville. And I'm a big Rod Stewart fan from way back!"

Talking to us in the quiet bustle of her New York office, Baran is articulate, philosophical, and seems content; her laid-back attitude is an eye of calm in the music industry whirlwind. Her approach to the biz is tempered by a respect for mundane professionalism, which she illustrates by describing the street sweepers of Paris: "When I see them wearing those clean, pretty, blue uniforms, sweeping the gutters with twig brooms, they look like they have a pride in everyday work that isn't filled with hype, pressure, or overextended ambition. There's a comfort," she says, "a reality and ease, that comes from approaching your work that way."

EQ: *How would you characterize your production style?*

Baran: It's a style of working, rather than a sonic style. The pre-production stage, which is intensely conceptual, is very important to me. I spend a lot of time with an artist to find out their tastes, what records they've done, what they did or didn't like about them, how they see their careers, how they see this project in the context of a career. I like to talk with people at the label to see how they perceive the artist, the product, the marketing problems; not that I'll pander to that, but to get a sense of what the project is, should be, could be. There are areas that some artists want to be involved in, and others they're happy to have taken out of their hands, and I'm pretty comfortable in all of them. I've done enough engineering to be able to think that through, and I have enough of a musical background to figure out arrangements and song structure. What irritates me the most is to be typecast by the way Laurie's records sound, or the labels that can be attached to her musical idiom.

EQ: *Laurie Anderson's records, from one to the next, appear to have quite distinct production approaches. They seem to have become more production-heavy. Has your creative contribution increased during the time you've worked with her?*

Baran: No. I had at least as much input in the early days as on *Strange Angels*. I think the differences have been more a result of her interest in working with a lot of different musicians, rather than banging through all of the parts at home by herself. That's the way she started. When she used other musicians, it was just one or two for flavor, not this large, ungainly orchestra of everyone in the world. The attempt to integrate performances by people who played months apart, in different studios, never having met each other, was one of the challenges of this record. It became a matter of, "What can we possibly take away?" rather than, "What can we cram in here?"

EQ: *Making* Strange Angels *involved 55 musicians, 15 studios, seven producers, ten engineers, and four mixdown engineers. How do you make a record under those circumstances?*

Baran: Slowly. It's not the most efficient way to work, but I don't think that was ever high on the list of criteria. When I came into the project, nearly all the other producers had already worked on one track or another. I was the last. I worked on it for over a year. Five or six of the tracks had been started or worked on a lot. Three or four were re-done from scratch or started fresh. A lot of good work had already been done by other people, so it was a matter of evaluating the song, the work that had been done, and Laurie's sense of what she wasn't getting out of what was already on tape.

EQ: *Did she switch producers out of dissatisfaction, or a desire to work with all of those people on one album?*

Baran: Some people came onto the project only to do one thing. But the real point about *Strange Angels* is that an artist's material—whether it's a demo, a song played on guitar, or 48 unmixed tracks—is just raw material. There's always something great about it and something not so great, problems whose solutions aren't immediately obvious, questions about the concept.

Also, there were some strengths in the music that, because she had heard it over and over, she wasn't able to enjoy anymore, but which I could appreciate. I would hear something and say, "That's great! Why would you want to throw that out?" I don't think I approached it any differently than I would any other project.

Sometimes I hear demos done on a Portastudio, but lovingly, over months, in a state of mind that's very different from the studio situation. The artist is alone and relaxed, and can be as critical and intensely emotional as he or she can. You can sit and punch in to a guitar track 400 times over a period of six weeks, because the time is free. Nobody's judging, criticizing, rushing. There's often something so incredibly charming about these demos that it's a big challenge to know what to keep, what to try to imitate, what to redo and recapture. Psychologically, it feels a little different to see that it's spread out over 48 tracks and has already cost 'n'-thousand dollars. But ultimately *Strange Angels* wasn't that different [from a demo].

EQ: *From one moment to the next, Laurie Anderson might speak, sing, or whisper. It must be difficult to capture her voice on tape.*

Baran: That was the string I tied around my finger and tried to keep in mind during all the mixes: To make sure the vocal was where it should be, that you could hear the words, that it was present enough. It's always a struggle. At one extreme, if you make sure you can hear every syllable regardless of the listening conditions, you get an oration declaimed on top of a practically inaudible music track. The opposite end is a completely buried heavy-metal screech, in which the vocal is just a texture and you don't want to hear the words anyway, because there aren't any. Somewhere along that continuum is the optimal place, but it's tricky. It's an interactive thing, spatially, sonically, in terms of EQ, everything. To pick up the phone and hear someone say, "I really like the Laurie Anderson record, but I couldn't really understand any of the words " -ugh! I don't want to hear that!

EQ: *Have you been dissatisfied with the vocal sound on her previous records?*

Baran: Not that I've been dissatisfied, but there's always a way to get closer, more focused, more intimate, more varied. Some of it is performance, some is recording. We've tried recording her under lots of different circumstances, but we always come back to recording at her place [a Manhattan loft containing "The Lobby," a fully equipped studio], because there we can take the time to focus, not on technical issues, but on the range and type of her performance—not pitch or inflection, but the more difficult abstractions of what the words mean and how they should be articulated. Should it be quirky? Withheld? Impersonal? Should it change from section to section? Understanding all the possibilities and having control over them is the hard, time-consuming part. Then, getting the pitch right and all that stuff is easy.

EQ: *Do you have special techniques for getting that clarity and definition in the tracks?*

Baran: We use a real traditional approach, a UREI LA2A limiter and a Neumann U89 condenser mike for the entire album. We use a tiny bit of compression at each stage—as it's going down, when we bounce together the composite vocal track, and during the mix.

EQ: *Much of the arrangement for the cut "Monkey's Paw" on* Strange Angels *seems to have been created in the mix from a chaotic multitude of tracks.*

Baran: Well, there was lot of percussion all over the place. We were trying to figure out militantly *laissez-faire*. Terry Roche was doing 'Joy To The World,' and she told me, 'I've been fooling around with it on my electric using this cheesy little amp, because that's what I have at home. But my sisters keep saying it sounds horrible!' I hadn't heard it, but I asked how she would do it otherwise. She said, 'On acoustic steel string, folk guitar, you know?' Then I asked what she'd *like* to do, and she said, 'I'm enjoying it with the electric. It doesn't sound bad to me, it sounds kind of exciting.' So I said, 'Why don't we do it that way then?' It was very interesting for once to encourage people *not* to think about what might be commercial. Some people did things that are very much in their style, but even then they turned out kind of quirky. I like the feeling that everyone sat around in their living rooms, playing in their underwear."

Laurie Anderson, *Strange Angels*, 1989

"The track 'Monkey's Paw' had been started before I came in, but it still went through changes. You can see from the way the song is structured on the record that it's quite sectional. Besides verses and choruses, there are little bridge-like events, an instrumental solo, an intro, an outro—it doesn't have the feeling of being through-composed, as some of the other tracks do. There were even more little sections, but at a late stage they ended up on the floor because the song was just too long and too fragmented—we lost focus. That also speaks to the questions of the different vocal sounds, because Laurie's 'personae' are different from section to section. The song starts with the first-person feeling: 'I went to the body shop, and I and I and I.' The chorus is more abstract and reflective, not 'I, I, I,' but 'the gift of life,' like a narrator commenting. Then there are some more bizarre elements, like the man who lost his head. And at the ends of the choruses, when she says, 'Haw, haw, haw, it's the monkey's paw,' there's a kind of scary effect—another, sort of sci-fi persona, coming from some other space with a more mysterious, satirically menacing edge. We recorded all the vocal tracks dry, and then processed them during mixing, but we gave some thought ahead of time as to what direction they would go in the mixing stage." •

what worked; thinning it out, fixing things—endless work done on the percussion tracks. The original bass track didn't work very well, so this bass player Bakithi Khumalo, who played on the *Graceland* record, came in late and re-did the bass part, which really pulled it together rhythmically. The guitar part was mostly there, but there were numerous tracks of it, with sections that worked in one place and didn't in others. We did a lot of moving things by sampling a bit from the third chorus to replace something that didn't work in the first chorus, to make the performances more coherent. It was mixed and re-mixed, and the final mix was done by Bob Clearmountain. He did an amazing job of pulling it together.

EQ: *What's it like to work with Bob Clearmountain?*

Baran: He doesn't articulate what he's doing, probably because he doesn't want to spend his time talking about his theory of mixing. We left him alone a lot and only interacted when something didn't seem to be working. Even then, we'd both strain to say, "What if you tried this?" And he would say, "Hmm." We'd come back later and he would have done something entirely different, but perfect. It was a thrill working with people of the caliber of Bob and Neil [Dorfsman, who mixed several of the tracks]. It allows you to keep your distance. If you have to be in there for 16 hours listening to every little EQ move and every variation in the settings on the outboard gear, you start to go numb. It's wonderful to work with someone you trust. All the hype about them is true. They're great.

EQ: *How do you keep up with new equipment?*

Baran: I'm in a lot of studios, and they always have a new something-or-other, so I make a point of hooking it up if I've never used it before. It's one thing to test something at a show or in a showroom, but it's quite another to use it on a track that you're familiar with.

EQ: *David Van Tieghem's album* Strange Cargo *is one of the most technology-intensive records you've been involved with. Besides producing it, were you involved as an arranger and as a composer?*

Baran: Yeah, although I wouldn't say I was extremely active at that stage. It's not my style. If I feel I'll have to interfere very much with an artist, I'm probably not going to work with that artist. Which is not to say I don't have strong opinions. There's a great image someone once used, comparing a producer with a male dog who has to piss on every fire hydrant and lamp post to mark off his territory—having an opinion whether one is necessary or not, changing things that don't need changing, fixing things that aren't broken—I'm real wary of being that kind of producer.

EQ: *Did you participate in the construction*

PRODUCER'S TIP

Consider putting down a track that you want people to play to, but don't plan to include in the final mix. It could be for feel, or to keep people doing overdubs from getting lost in the murk, or to strengthen certain sections. Putting down a fast-moving hi-hat part, so there's a sense of an internal subdivision of the beat, might encourage a guitar player to play more lightly, or to use some kind of motion that feels the same way. A gigantic fill—one so hideous you know you'd never use it, but which makes it clear that something big is about to happen—can be very helpful to signal to everyone that, say, the chorus is coming. Of course, sometimes it happens spontaneously. You just pull something out of the mix, and there's that nice feeling that it's missing, but it still holds the other parts together.

—Roma Baran

of the synthesized and sampled sounds?
Baran: Sure. It's more fun to work together. At certain moments I'm standing back and expressing opinions on a larger scale—that's called production—and there are moments when it's more like, "That's fun, let's do this"—just two people screwing around. That's production, too. But I try to keep some distance, because if I become too involved in the actual procedures, I get into this low-level, wirehead state of mind: I'll try to do a real good job, focusing on what's right up against my face, and four hours later I'll look up and think, "Wait a minute. Shouldn't we have printed the guitar three hours ago?" I lose the larger perspective. I don't play on projects [any more] for the same reason.
EQ: Strange Cargo's *liner notes mention the Apple Macintosh, Opcode's Performer and Vision sequencing software, and Digidesign's Sound Tools disk-based recording system.*
Baran: I think of the sequencer more as a tape recorder than as a "mechanical" device. It's there to capture a performance, rather than to make something perfect. If I'm going to quantize, I think very carefully about it first. I do like to use sequencers for subtly changing the tempo of a song; the kind of tempo changes a human being does naturally, even if the overall performance is mechanical.

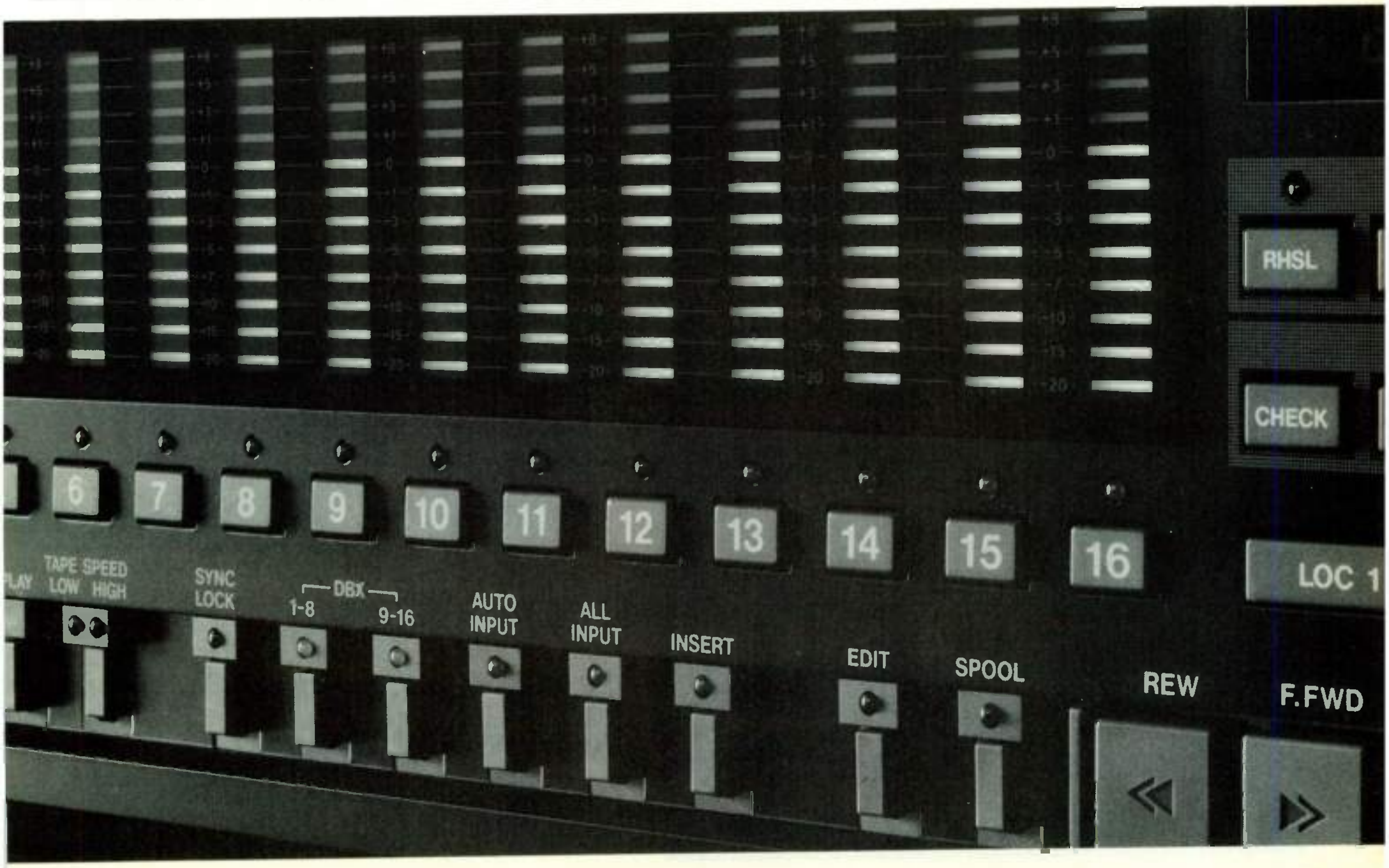
I've used other disk-based systems, but Sound Tools is gaining popularity because of the ease with which a facility can set it up—the price, friendliness, and compatibility with other formats. Before, you might have had to go to a Sony digital editor to accomplish similar things.

For Laurie's record we used the Sony 3000 digital editor at Masterdisk [in Manhattan] to do the 2-track editing and assembly. The 3000 is a pleasure to use because it's so well-organized visually. There are two little windows, and it shows you a little section of memory. The incoming edit comes up—you see it in real time—and you see the crossover point and the outgoing section.
EQ: *Were you simply putting together the master, or did it involve editing out a verse or a chorus?*
Baran: There were some internal edits.
EQ: *It seems risky to save internal edits until that late in the process.*
Baran: Absolutely. It's not like we saved them for last; at a late stage in the process we said, "Oh no, we've really got to cut this verse out." With material that's more repetitive and predictable, like dance and house music, it's easier to say, "We're going to shuffle this so the form is ABA instead of ABBA," or "We're going to clip this piece out and use it as an intro." With material that has a more complicated shape, it's important to be hearing it in its final structural form early on. On *Strange Angels*, there were a few songs in which we did 48-track, 2" edits. That's a serious pain. Each edit takes four hours, and something always goes wrong. It's always really stressful

PRODUCER'S TIP

I'm not an advocate of printing everything dry and clean so you can process it later. Except for vocal tracks, I make an effort to print effects to multitrack—not basic reverb, but musically integral effects. A lot of the music I work on is so dense that you constantly need a context. If you're listening to 11 tracks and none sound anything like what they're going to sound like in the end, how are you supposed to make a decision about the 12th? If you expect to put a triplet delay on the woodblock and a two-second reverb on power chords, then why not print them? Of course, it's dangerous. But generally it's a better way to go.

—*Roma Baran*



because you have to make sure you don't end up losing your lock-up.

EQ: *Would you describe the procedure for making a viable 48-track analog edit?*

Baran: You have two reels of 2" tape, each with SMPTE time code on track 24 and click on track 1. If you cut the two pieces of tape on both sides and put them back together, the code would be discontinuous, and at the edit point, the machines would lose their lock-up. So you create a free track on each reel—not always easy at a late stage in the production—and put a reshaped copy of the original time code on it.

Then you use the reshaped code to jam the code back onto track 24, across the edit. You need to lock up to the internal crystal of the synchronizer while you're jamming, because if you just jam from one track to another on one machine and one 2" tape, the tiny variations in speed that are created when you do the jam are not reflected in the tiny variations that occur when you do it on the other reel. So you need some other source, which is the crystal in the synchronizer. That way, the new code on both reels is referenced to the same sync source.

But there's always some kind of problem. There may be little blips in the front when you're jamming, or you've added an intro to the song and now there's not enough pre-roll, or the physical cut in the tape causes a problem even though the jammed code is continuous. Sometimes we would do absolutely everything right, and we'd test the lock-up across the edit by listening to the click in stereo—the click from the master on the right, from the slave on the left—and it would be perfectly phase-locked right up to the edit. And then you'd hear it stutter. It's such a slippery process that even if you have a bunch of people who know what they're doing, there are always two or three problems.

EQ: *Under what circumstances do you decide it's necessary to do something as drastic as cutting across 48 tracks?*

Baran: If the mix is complicated, and the song's dynamic shape is mysterious enough that it would throw off the process to keep stopping at the end of, say, chorus 2 and doing the rest as though it were a separate mix. Unless you keep stopping and doing test edits—listening to the mix by printing the two pieces and cutting them together several times as you go along—you never really get to hear the continuity over and over again. Instead, for hours you're listening to the first half of the song and then the second half of the song, separated by some amount of dead space.

PRODUCER'S TIP

I like to get most of the way through a mix before I bring in musicians for overdubs. That way, they're performing to something that sounds more like a real song. I do the same thing with vocalists and back-up vocalists to get them to feel the spirit of a song. I get better performances that way.

—Roma Baran

Cutting 48 tracks of analog 2" is something you don't want to make a habit of.

EQ: *As the technology becomes more sophisticated, how do you reconcile technology and creativity?*

Baran: That's part and parcel of the kinds of compromises you need to make as a producer. The technology has developed to the point that the degree of perfection you can achieve is almost limitless. You can quantize, move around notes to within 1/480 of a quarter-note, fiddle, diddle, do and re-do. It's like all innovations: a blessing and a curse. It's hard to know how much you can afford, in terms of your energy and focus. You have to assess the damage you're doing to the freshness of the material. •

A ROOM EQUALIZED TO PERFECTION

MULTITRACK TO MIXDOWN

STUDIO MONITORS REVEAL

THE ENTIRE SOUND SPECTRUM

Now the excellence of sound experienced by engineers and producers can be duplicated on Sunkyong Pure Chrome Tape.

What the producer hears, chrome remembers.

UCR Pure Chrome Professional Duplicating Tape from Sunkyong.

Because the time has come for the world to share the excitement of studio sound.

SUNKYONG

Professional Products

Sunkyong Magnetic/America, Inc., 4041 Via Oro Avenue, Long Beach, CA 90810 1-800-331-5729 Outside CA 1-800-237-8372

Both methods record incredible stereo. Crown's SASS™-P makes it fast and easy.

Crown's new Stereo Ambient Sampling System™ (SASS™) and the DAT recorder combine to create a new generation of stereo recording. Unlike traditional recording methods, this powerful new combination is easy to use, more dependable and less expensive. The result is a system that records natural, beautifully imaged stereo just as your ears would hear it.

Crown's SASS-P* eliminates traditional stereo recording compromises in sound quality, ease-of-use, and cost. No longer do you have to settle for weak low-end or off-axis coloration common to Midside, X–Y and near-coincident pair mics. Assembly and positioning time is also reduced significantly compared with conventional stereo micing techniques.

*Also available as SASS-B for use with B&K 4003/4006 microphones (not supplied).

Lightweight and easy to set up, the SASS/DAT combination is a high-quality portable recording system that goes everywhere.

The SASS-P features switchable battery or phantom power and Crown's finest studio-grade PZM® capsules. You'll enjoy full ambience without coloration, excellent sum to mono, and extraordinary broad frequency response. With SASS's superb imaging capabilities, every sound is audibly reproduced in its precise position, resulting in a stereo experience of uncanny realism.

Readily adaptable to all common stands, the SASS includes a carrying case and accessories.

No matter what your stereo recording requirements—from sampling to electronic news gathering to remote recording of live events—you'll find the SASS-P the clear choice.

See your Crown representative for more information or call toll-free: 1-800-535-6289.

SASS Sample CD!

Now you can hear for yourself just how great SASS sounds. This CD includes a wide selection of live and studio music, stereo sound effects and samples, including A-B comparisons with some of the best stereo mics available. The SASS CD is free, but we must charge $4.95 for shipping and handling charges. Send check or money order to: Crown SASS CD, P.O. Box 1000, Elkhart, IN 46515-1000. Please allow 3-4 weeks for delivery.

Name ____________________
Title ____________________
Company ____________________
Street ____________________
City ____________ State ______ ZIP ______
Phone ____________________

crown®

Crown International, Inc. • P.O. Box 1000
Elkhart, Indiana 46515-1000 • 219/294-8000

IN THE POCKET

Tax Talk: Options For Business Organization

BY STEVE HARLAN

IN THE MAY/JUNE ISSUE OF *EQ*, WE discussed the pros and cons of organizing your studio business as a sole proprietorship or a partnership. Now let's take a look at some of the other options.

The Corporate Way. As you might expect, the form of doing business that provides the greatest potential tax (and non-tax) benefits also is the most costly and difficult to administer. Legal fees, for instance, for even the simplest business incorporations, range from $600 to $1,000. Yet the benefits can far outweigh the costs.

For tax purposes only, there are two types of corporations: "S" corporations and "C" corporations. (Poetic, huh?) Notice I said, "for tax purposes only"; when it comes to non-tax matters, the "S" and "C" distinctions are meaningless. An "S" corporation is one that has elected to be taxed as a partnership. A "C" corporation is any corporation that is not an "S" corporation. Non-tax benefits, such as *limited liability*, apply equally to both. Limited liability is good: It means that even if the corporation goes bankrupt, the owner's personal assets and credit rating are not harmed.

	Sole Proprietorship (Individual)	"C" Corporation Individual	"C" Corporation Corporation	"C" Corporation TOTAL
Income	$100,000	$50,000	$50,000	$100,000
Deductions	$20,000	$20,000	N/A	$20,000
Taxable	$80,000	$30,000	$50,000	$80,000
TAX	$21,744	$5,996	$7,500	$13,496

"C" Corporation. A big tax advantage of the "C" structure is known as "income splitting." Let's say you're a fairly successful studio owner operating as a sole proprietorship. Your studio nets you a cool $100,000 annually. You file your taxes as a single individual and you have personal deductions, such as mortgage interest, medical expenses, dependents, and so on, adding up to $20,000 per year. Let's assume that you only need a $50,000 annual salary to make ends meet. You learn from your accountant that corporations pay only 15% tax on the first $50,000 of income, so you decide to incorporate and have the corporation pay you a salary of $50,000. Let's compare the outcome (refer to the chart).

As you can see, the incorporation saves $8,248 ($21,744-$13,496) annually.

Other tax advantages available to a "C" corporation are tax-free benefits such as qualified medical reimbursement plans, disability plans, and retirement plans. Only a "C" corporation owner can borrow from his or her retirement plan.

The *disadvantages* of income splitting are complicated in theory, so they're beyond the scope of this column. Basically, they include: accumulated earnings tax; double taxation on liquidation or sale; and 34% tax rate on Personal Service Corporations. Discuss these issues with your accountant before incorporating.

Tax Hint: *Don't set up a "C" corporation for a new business that is going to show losses for the initial operating period. Those losses may be used only to offset future "C" corporation income, if any. Instead, elect "S" status for the corporation so the losses "flow-through" to the owners—this means they are deductible on the partners' individual tax returns.*

"S" Corporation. This form of business is taxed, more or less, as a partnership. One important advantage is that the income after reasonable salaries to owners is *not* subject to the 15.3% self-employment social security tax. I'll discuss that in detail in a future column.

The "S" corporation is bound by the same restrictions on retirement plans and fringe benefits as a partnership. Since "S" status is elective, however, when these aspects become a problem, the owners simply elect out of "S" status. *Once you have elected out of "S" status, you cannot re-elect "S" status for five years.*

Thus the "S" corporation provides you with partnership tax treatment, a potential reprieve from social security taxes, and limited liability.

Limited Partnerships. Although the use of limited partnerships is fairly rare in small businesses, I should point out that they can be a great way for the owners (general partners) of a studio to attract investment capital from investors (limited partners) who will not actively participate in the business.

> If You Want Partnership Tax Treatment, Possible Reprieve From S.S. Taxes, And Limited Liability, Say "S."

Bottom Line. Making the right choice about the form in which your studio or music/production business will operate is crucial. Ultimately, you must weigh the costs involved in operating in a more sophisticated form against overall tax and non-tax advantages. •

Steve Harlan is a certified public accountant, musician, and studio owner based in Portland, Oregon.

TECH SUPPORT

STUDIO CLINIC

Recording Levels & The State Of Flux

BY RICHIE MOORE, PH.D.

MOST OF US ENGINEERS ARE USED TO seeing cryptic notes such as *30 ips, 456, ND,* and *+3dB* written on the session setup sheet. These simply tell us that we should set up the tape machine to record at 30 inches per second, use Ampex 456 tape, record material without noise reduction ("*No Dolby*"), and calibrate record and playback levels so that 0VU equals +3dB. Simple, right?

Well, not exactly. What about this playback and recording level figure of *+3dB*? Specifically, this means that the tape has been recorded slightly "hot"—so that a 0VU reading on the tape deck is in fact 3dB above the nominal operating level of the deck (which is +4dBm on pro decks or -10dBV on "semi-pro" decks). But to calibrate a deck to "+3" or any other setting, you need to be familiar with the test tapes that provide the reference tones necessary to adjust the deck for playback levels.

Playback calibration is one of the most critical adjustments you'll make before beginning a session, and it must be done properly because record levels are referenced to the tape machine's playback section.

The Technical Side. Since tape recorder heads are magnetic devices, measurements and calibrations are based on their magnetic fields. Magnetic *flux* is a measure of the strength of a magnetic field; the amount of flux in a device is called its *flux density*, or *fluxivity*. Fluxivity is expressed in *Webers* (Wb). (Without getting into too much rocket science, one Weber produces 1V in a one-turn conductor over one second.)

The Weber is a large value—conceivably large enough to power an extraterrestrial spacecraft of some sort. Magnetic tape, on the other hand, can be affected by devices whose fluxivity is 10^{-9}Wb—one billionth of a Weber. Consequently, when we measure tape recorder reference fluxivity we use the term *nanowebers* or *nWb* ("nano" = billionth) *per meter* (of tape).

Reference Tapes. The basis of reference fluxivity dates back to the early Ampex tape recorders. Ampex set their 0VU playback reference level fluxivity to 185 nanowebers per meter (nWb/m) at 700Hz. Since Ampex was the original pro tape recorder manufacturer in the United States, this reference became the de facto standard.

Standard Flux Levels
(referenced to 185nWb/m)

185nWb/m	= 0.0dB
200nWb/m	= +0.7dB
250nWb/m	= +2.6dB
259.6nWb/m	= +3.0dB
320nWb/m	= +4.8dB
360nWb/m	= +5.8dB
370nWb/m	= +6.0dB

Higher flux densities increase signal-to-noise ratio, but at a price.

The '70s saw the development of low-noise high-output tapes, on which hotter signals could be printed. (The first of these tapes, Scotch 206 and 207, appeared in late 1969, followed by Ampex 456, Scotch 250 and 226, and Agfa 468 and 469, to name a few.) Prior to the advent of reference standard tapes for use with these new high-output tapes, you took your trusty 185nWb/m test tape and set the playback level on the tape machine to read -3dB VU instead of 0VU. Then you would calibrate your record levels so the test tones would read 0VU. By this wonderful sleight of hand, you were able to increase the signal-to-noise ratio by 3dB.

That's why the industry came to create test tapes with formats specifically designed for use with the new generation of recording tapes.

One of the most popular test tapes in modern recording studios is the 250nWb/m test tape, or +3dB over 185 nWb/m. (A 250nWb/m test tape, however, is not really +3dB. It's actually about +2.6dB. Another popular test tape is 200nWb/m, also referred to as +1dB over 185. To further confuse things, this is actually +0.7dB above 185. But for general audio purposes we round off these two figures to +3dB and +1dB, respectively.)

Two major manufacturers of calibration test tapes are Standard Tape Laboratories (STL) in San Leandro, California, and Magnetic Reference Laboratories (MRL) in Mountain View, California. Both companies' tapes conform to the IEC standard "Magnetic Tape Recording and Reproducing System: Calibration Tapes," and their equalizations conform to the standards published by the NAB, AES, and IEC.

Some fundamental differences do exist, however, between these two companies' reference tapes. STL uses 700Hz as the reference frequency, and comes with reference flux densities of 185nWb/m and 259.6nWb/m (true +3dB over 185nWb/m). MRL uses a 1kHz reference frequency and comes in 200, 250, and 320nWb/m formats. The 320nWb/m is made according to European DIN standards.

Aligning the playback electronics of a tape machine to higher reference flux densities lets you increase the signal-to-noise ratio by as much as 6dB—but you can run into problems. You must make sure that peak signals do not exceed 0VU; otherwise, you may run out of headroom or reach the point where the tape becomes saturated, which results in increased distortion and compression of the signal. Another pitfall: The hotter you record on tape, the greater the potential for print-through from one layer of the tape to another.

It's a matter of individual preference—but I think that true +3dB over 185nWb/m provides the optimum level. •

Richie Moore, PhD, is chief technical guru at the Plant Recording Studios in Sausalito, California.

FILM & VIDEO SOUND

Where Can I Find Bott's Dots?

BY LARRY BLAKE

THE TASK OF ORGANIZING A SOUND effects library presents some fairly intimidating questions, such as: Should we divide the tapes and disks by subject, project, or in the order recorded? Should we cross-reference with 3x5 cards, notebooks, or computer? Should we identify vehicle sounds as "auto" or "car"? Should we put those sounds in the same place as those of trucks and motorcycles?

I highly recommend shelving and editing serious libraries according to subject matter. This is expensive, due to the time needed to do it properly. However, the investment pays off in the future, when you will have to shuffle DAT cassettes or optical discs to audition sounds in your digital editing system. The fast access time of optical discs will be offset if you have to insert 20 discs to hear 20 effects. Organizing sounds by subject drastically cuts the need to swap.

Some Suggestions For Organizing Sound Effects, Present & Future.

When you organize your library by subject, you have to determine your subject divisions: Are "ricos" (ricochets) filed under "guns" or do they represent a separate category? Ditto for tire peel-outs and autos. Keep in mind that inconsistencies in descriptions make it much harder for someone unfamiliar with your library to find a given effect.

If you organize your effects on a computer, you still have to decide how to structure your database (assuming you don't buy an off-the-shelf sound library program), and how to cross-reference. This is no simple task, and requires a great amount of planning.

In the last issue of *EQ*, I wrote that I haven't seen any digital editing system seriously address the issue of accessing sounds via computer database. Now, through the auspices of the Society of Motion Picture & Television Engineers (SMPTE), I'm chairing a committee that will develop a recommended standard for sound library databases.

The committee's task is two-fold. First we will create a glossary of sound effect descriptions. At the same time we will divide all effects into logical groups of categories and sub-categories. We'll develop a numerical indexing system, which will help in cross-referencing of effects, and in the physical organization of libraries. Numbers leap across language barriers, and a cross-reference indexing number lets someone accomplish complex, multiple key-word searches quickly and efficiently.

Our second goal is to define the structure, length, and contents of the computer database fields. Our design will not be for any specific hardware or software. We intend for it to take into account the limitations of inexpensive filing programs, without handicapping the power of personal computers. If manufacturers conform to the recommended database field structure, you will be able to integrate commercial sound-effects libraries into your private library, with minimal data-translation headaches.

I also hope that the committee will publish its findings in print *and* as a comma-delimited computer file. The latter will let manufacturers of digital sound-editing products use our glossary and numerical indexing system in an on-line help function that could be part of the key-word search through the database.

Say you want to find out if the library has the sound of tires rolling over those thingamajigs on highways that separate the lanes. You type in "lane marker," a good guess. Depending on its abilities, the computer can tell you that *it* calls lane markers "Bott's Dotts," and you have to type in "Bott's Dotts" (or just "Bot's Dots") to learn what cues are in the library. Or the computer can show you these cues, knowing that "lane marker" is a synonym for "Bott's Dotts."

A studio or production facility can adapt only those parts of our findings with which its engineers or editors agree. For example, although you may use the numerical indexing system to organize effects by subject, you can continue using your current system of organizing the tapes and disks themselves. And you also could use what probably will be our two most important recommendations: lexicon (word choices) and database field structure.

You don't have to follow *all* the recommendations word for word; you can call it "car" even if the recommendation is "auto." If you don't want to include all the fields specified in the database, don't. Our goal is to present a comprehensive and well-planned starting point.

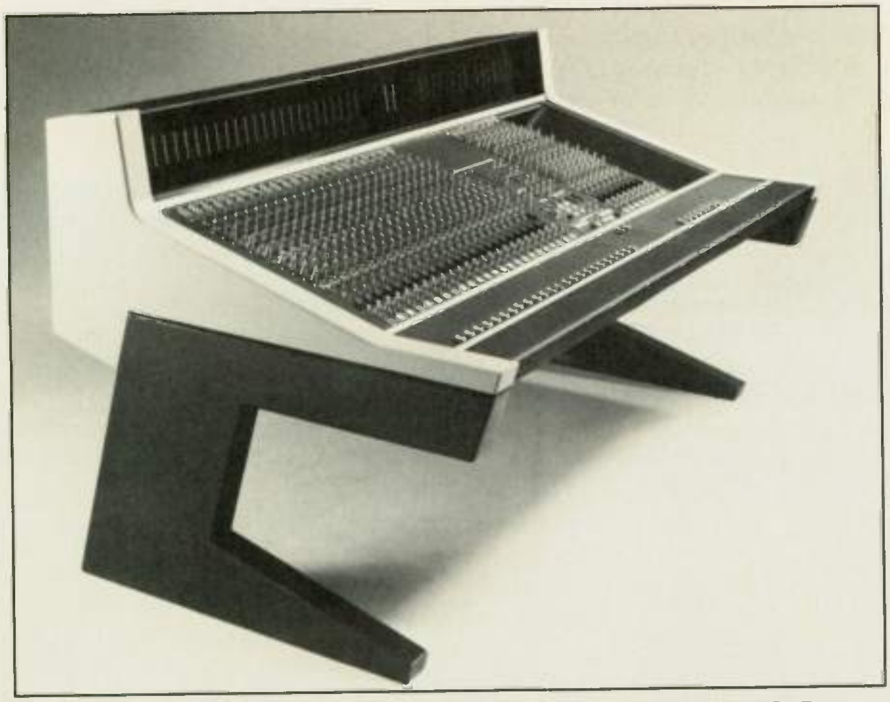
On the other hand, because we're aiming for the deep center-field stands, our findings will make almost every current method of organizing a sound library "obsolete." Again and again we hear, "Do you mean I will have to . . . ?" The answer is, we take our design cues from existing libraries, and seek common threads in the way people organize sound effects. However, since there are few similarities among companies that sell sound effects, what we recommend will not resemble any existing system.

The committee's work is far from complete, and we are open to suggestions. If you want to see our current drafts (we haven't yet submitted any recommendations through SMPTE's long standardization process), please write to me at Box 288, Hollywood, CA 90078. •

Larry Blake is a sound editor and re-recording mixer based at Weddington Productions in North Hollywood, California.

BE QUIET!

If you assign all your channels to the stereo mix buss with the channel faders, master faders, and your control room monitors wide open and you hear more than just small hiss, maybe it's time to change mixing consoles. Get a console with specs and sound you don't have to apologize for. BE QUIET with a new D&R mixing console and hear quiet like you've never heard before. Call our customer support department at **(409) 588-3411** for a "factory direct" quote today.



Pictured above is the D&R AVALON which will premier with several new D&R consoles at the AES 89th Convention in Los Angeles, Sept. 21st thru 25th.

TOTAL CONTROL

D&R

mixing consoles

D&R USA • Rt. 3, Box 184-A • Montgomery, TX 77356 • Phone: (409) 588-3411
D&R Electronica b.v. • Rijnkade 15B • 1382GS Weesp • The Netherlands • Phone: (--) 31 2940 18014

Confused About "Exciters"? Read the Facts.

Seems like a good thing always leads to imitators. Which is why there seems to be a rash of so-called "brightness enhancers," "phase correctors" and "exciters."

The Aphex Aural Exciter® is a *patented* audio process that will recreate and restore missing harmonics. When added, they restore *natural* brightness, clarity and presence, and actually extend audio bandwidth. All without adding any appreciable power to the signal.

As a result, the Aural Exciter has become a standard tool in the recording, film, broadcast and sound reinforcement industries around the world. It has been licensed for use in products sold by Yamaha, Numark, AKG, Proton, Gentner Engineering, MacKenzie Labs, and Vestax.

Other "brightness enhancers" only boost existing high frequencies, pumping as much as an additional 12dB, which can distort the amp or even blow your speakers... in addition to sounding unnatural. In fact, you could probably achieve the same effect more flexibly and economically by using any equalizer.

Don't be confused by hype. Listen to any device claiming to do what only an Aphex Aural Exciter does, then listen to the real thing. Your ears will hear the difference.

APHEX
SYSTEMS

All Aphex products are designed and manufactured in the U.S.A.

11068 Randall Street
Sun Valley, CA 91352
(818) 767-2929

LIFE AT THE EDGE OF THE BOOM

MICROPHONE TECHNIQUES

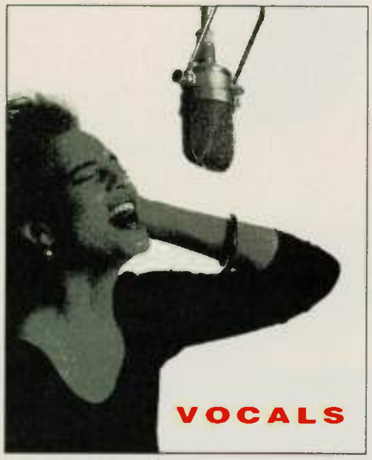
EQ WORKSHOP

REMEMBER MICROphones? They're those funny-looking metal things that we used to use for recording instruments. Of course, that was before the synthesizer, MIDI, and virtual-track sequencers were invented. Still, a few

BY MICHAEL MARANS

PHOTO: FRED STIMSON

diehards insist on using live vocalists, horn sections, and, worst of all, real drummers. Seems the microphone still has a place in today's studio. Whether you're a seasoned veteran wanting to brush up, or a MIDI maven new to recording through the air, we invite you to partake in our microphone workshop. You never know when some wide-eyed technophile will get a bit crazy and decide that one of his cuts just won't be complete without an acoustic guitar track.



ALTHOUGH NEARLY ALL PROFESSIONAL vocal recordings are made using large-diaphragm condenser mikes, there are no hard and fast rules for choosing the microphone. Even an untrained ear can detect nuances in the frequencies in the vocal range; accurate reproduction of these subtleties translates into emotional impact. The paramount consideration is the timbral quality of the voice you're recording.

Two of the most popular vocal mikes are the Neumann U87 and the AKG C-414. The Neumann usually is chosen for its warm sound and midrange resolution, and the AKG for its 10kHz "sheen." Dave Frazer, engineer at Narada Michael Walden's Tarpan Studios in San Rafael, California, relies almost exclusively on his modified (by Klaus Heyne) C-414. To record Whitney Houston's vocals on her debut album, he used a C-414 containing an older C-12-style capsule, running through a Focusrite preamp and equalizer and a Neve compressor. "The 414 is an excellent mike to start with, due to its presence," he says. "You do have to equalize some of the midrange out or it can be a bit too edgy, but it works for me almost every time."

As you might expect, engineers—and for that matter, vocalists—have their own preferences. Barbra Streisand uses a customized Neumann M49, Bette Midler, a Telefunken ELA M 251, and Aretha Franklin (and, on occasion, Michael Jackson), a large-diaphragm dynamic Shure SM7.

"In the pop music domain, you want to pick a microphone that gives you a lot of EQ flexibility; there usually isn't enough time to put up three or four mikes and listen for the best sound," says Joey Wolpert, who co-engineered Bette Midler's Grammy-winning "Wind Beneath My Wings." Wolpert believes that the AKG C-414 is not as forgiving of less accurate high-frequency equalization as the Telefunken

MICROPHONE BASICS

EQ Background Notes

CONDENSER vs. Dynamic. As a rule, featured instruments are recorded with condenser mikes. Like the eardrum, the diaphragms in condensers are so thin and light in mass that they can register even the finest changes in air pressure. Condensers are more subject to distortion, however, when recording source signals that have a high sound pressure level (SPL), such as an amplified guitar or kick drum. In these applications, we generally use dynamic mikes. The trade-off is that dynamic mikes don't offer the same subtlety of expression as condensers.

Small- vs. Large-Diaphragm Mikes. Mikes with small diaphragms (3/4" or less in diameter) frequently are preferred in situations that call for pure accuracy, quick transient response, and flat frequency response, such as classical recording. However, many engineers feel small-diaphragm mikes offer lower "sonic resolution" than large-diaphragm mikes, reflecting a traditional preference for the coloration characteristics associated with large-diaphragm mikes.

Each recording situation is different, but here's the traditional rule of thumb for many pop producers: Use a large-diaphragm microphone for the main vocal, or featured acoustic instrument; use a small-diaphragm mike for supporting instruments. This way, the featured instrument will have all the subtle color that a large-diaphragm mike provides, allowing it to stand out from the supporting instruments. Your supporting instruments will sound just fine, but they won't compete for space with the main instrument when it's time to mix. (You'll also find that small-diaphragm mikes won't have to be padded [attenuated] as much as large-diaphragm mikes when you're setting input levels.)

Use large-diaphragm mikes such as the Neumann U47 and AKG C-12 when a rich, warm sound is desired. Small-diaphragm mikes, such as the Neumann KM84 with its excellent transient response, are great for percussion instruments.

What Pattern Should You Use? Omnidirectional mikes are capable of reproducing the source signal with great accuracy, and in most cases the results are more pleasing aurally. Another advantage of omnidirectional microphones lies in their construction, which is relatively simple; high-quality omnidirectional mikes usually cost far less than equivalent directional models. In general, however, you only want to use them if your instruments are relatively isolated. Otherwise, you'll pick up unwanted signals through the back and sides of the mike.

Usually true isolation is

251, Neumann U47, or M49. "U87s tend to accept top-end EQ," he adds, "because they're not as bright to begin with."

Due to the time constraints in today's sessions, Wolpert is a big fan of Dolby SR noise reduction. "When we recorded Bette, Arif [Mardin, producer] gave me very little time to get a vocal sound. Having SR allowed me to do all of my EQ'ing after the vocal was recorded. SR gives you the luxury of having lots of headroom, and when you add EQ, you're not EQ'ing the tape hiss." [*Ed. Note: For specific recommendations see the accompanying "Mike Applications Chart," which lists polar patterns and recommended applications for over 50 models.*]

Mike Placement. First, try to find the spot in the room that has the smoothest frequency response and the least amount of noticeable decay. Try to avoid placing the microphone near flat, hard surfaces, as they can cause unwanted reflections.

Once you've found a place for the mike, it's time to position the singer. The distance between the vocalist and the microphone should be no less than 3" or 4", otherwise you run the risk of inducing mechanical distortion. Six inches is a good starting point for condenser mikes; if you're using a dynamic mike, the singer should stand or sit a bit closer to the mike to avoid losing low end.

Most engineers prefer to place the mike a bit above the vocalist, with the diaphragm pointed toward the mouth. The more directly the mike is aimed at the throat, however, the harsher the sound is, so it's best to aim a bit low or a bit high. Aiming high helps keep the vocalist's throat open, as he or she has to look up slightly; aiming low is good for vocalists who always look down to read lyric sheets.

Wolpert likes to put singers through a few "calisthenics" to find the "sweet spot." "I have the singer rise up on his or her toes and sing, then bend at the knees and sing, then move back and forth a few inches," he says. "This lets me find the spot where the 's' sounds round out, and helps me determine the 'size' of the vocal. You can get a vocal that's too big and it can overtake the track. The idea is to find the distance where you can get a vocal that's the right size while using the least amount of compression." What happens when the right spot results in the unwanted popping "p"? Wolpert replies, "I've always found that it's better to ask the vocalist to mind their plosives than to give up the sound of a sweet spot."

A few words about windscreens: Always use a stocking or a custom nylon windscreen—*no exceptions*. Don't use foam screens—they dramatically attenuate high frequencies.

Background Vocals. There are as many different ways to record background vocals as there are vocalists. For small groups (three to four singers), try something like a U87 or a C-414. Use cardioid mode to help focus the sound on the singers, rather than on the room. The mike should be far

not possible, hence the popularity of cardioid mikes. Unfortunately, when cardioid mikes are used, the sound quality is compromised due to minor phase cancellation caused by the directional pattern. When recording classical music, some consider this to be a problem, as you're generally trying to achieve a full-spectrum sound. In pop music, however, you're creating the spectrum, and you can manipulate a sound to fit into almost any frequency region. For example, 10kHz isn't naturally found in a singer's voice, but you can add that frequency to bring out the "airiness" in the vocal. Another notable characteristic of cardioid mikes is called "proximity effect"—a low-frequency boost that occurs when the mike is held very close to the sound source.

Other mike patterns include supercardioid, hypercardioid, and bidirectional. Supercardioid mikes, such as the EV RE15, Beyer M400, and others, offer the greatest unidirectionality, and are most effective at differentiating between sounds coming from the front hemisphere (as opposed to the rear).

Hypercardioid microphones, such as the Audio-Technica AT4053, Fostex M1109, or Beyer M69, work at a greater distance than standard unidirectional models. When recording in a setting where unwanted sounds are coming from all directions, these make a good choice.

Bidirectional ("figure eight") patterns normally are used for special applications, such as stereo miking. One of the best times to use bidirectional mikes is when working with two singers, who can record facing one another. The shortcoming of a figure-eight polar pattern, however, stems from the fact that it is not unidirectional, thus increasing the possibility of picking up unwanted sounds from other instruments.

Tubes vs. Transistors. Most engineers agree: There's nothing like the sound of a great tube mike. The problem is, the tube is the determining factor in the mike's sound; if the tube is bad, so too will be the sound. Rather than take a chance on a questionable tube, many engineers opt for the consistency offered by solid-state mikes.

Contrary to popular belief, the difference in sound between a tube mike and a solid-state mike is not due to the tubes or transistors. According to microphone expert Klaus Heyne, the difference is attributable to the mike's circuit design. Usually the circuit in a transistorized mike is more complex than that of a tube model, often due to the extra features, such as dB cut, bass roll-off, and variable polar patterns offered by solid-state mikes. Klaus' clients frequently ask him to modify transistor mikes so they sound more like tubes; he starts by eliminating all extraneous circuitry.

enough from the group so that the people on the sides are picked up; if all the singers can see the face of the mike comfortably, it's in the right place.

For larger groups (six to eight people), try omni mode with the singers circled around the mike. Omnidirectionals such as B&K's 4006 work well, especially if the singers vary in height. Place the mike at the group's average height, about 2' away.

You also can record groups of four or more using a stereo "A/B" type of placement. Place two mikes about 6" apart and about 4' away from the group. Use a cardioid to maintain the stereo separation.

BEFORE YOU DECIDE HOW TO MIKE ANY acoustic guitar, you have to *listen*. Go into the studio and ask the guitarist to play for you, so you can hear the natural sound of that particular guitar. Admittedly, many times you won't be aiming for a natural sound. But even if you know you're going to put gobs of compression on an acoustic guitar track, it helps to know what the instrument sounds like in the first place.

Several accepted methods apply to recording acoustic guitar, each involving two microphones. The first mike is used primarily to record the sharp attack of the pick plus whatever string and fret noise is present. Suggested mikes include condensers such as a Neumann U47FET, AKG C-414, and AKG C-452. Most engineers agree that it's desirable to use this close-up mike in omni mode when the room sounds good, but most end up choosing a cardioid pattern since omni usually provides too much information.

Place this first mike 6" to 10" away from the front of the guitar (the closer the mike, the more pick noise you'll hear), on the neck side of the soundhole. If the mike is situated on the bridge side of the hole, there's a good chance the sound will be too tinny. Point the mike slightly toward the neck of the guitar (although interesting timbres can be obtained when you point it more toward the soundhole). Either way, make sure you don't point it directly at the soundhole. A low pulse of air comes out of the soundhole, and this produces a boomy, unbalanced sound.

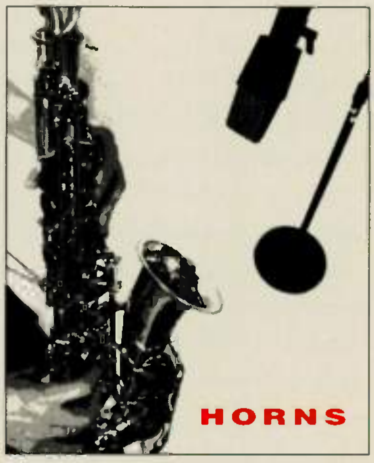
Whenever boominess is a problem, back the mike off. Boominess is not the only problem of close-miking an instrument with a cardioid microphone: The slightest movement of the instrument induces the proximity effect, producing a change in frequency response. If the problem won't go away, try an omnidirectional microphone. Also try positions up to 3' away to capture the full tonal range of a fine acoustic.

Another caution: When you record acoustic instruments that have flat surfaces, such as a guitar soundboard, avoid placing the mike in such a way that the plane of the diaphragm is parallel to the plane of the instrument's surface—this can result in standing waves. Tilting the mike slightly off axis results in a more even decay and more defraction, and *that* means you'll get a fuller sound.

The second mike is placed farther away than the first, to pick up more of the actual tone and body of the guitar. Here again, you have placement options. One method is to use a large-diaphragm condenser microphone placed about 3' away, aimed at the guitar body. When miking a solo steel-string flat-top or nylon-string classical, consider using highly accurate small-diaphragm condensers. A less common technique involves using a shotgun microphone such as Sennheiser's 816, as far as 8' or 9' back from the instrument.

It's important to be aware of potential phase-correlation problems when using two-mike techniques. *Always* test for phase problems by monitoring the signals from the two mikes in mono, then altering the phase of one of the mikes. When you find the setting that results in a full-bodied sound with good low end (phase problems tend to manifest themselves as a decrease in the low frequencies), you've found the proper setting.

Your overall guitar sound is determined by the blend of the near and far microphones. In general, if the guitar is the featured instrument, you'll want to have more signal from the far mike, as this is the mike that provides the body of the tone. If the guitar is used for multitrack overdubs, the near mike should be dominant.



WHAT MICROPHONE WOULD YOU USE to record Miles Davis' trumpet? Common sense tells you that a trumpet puts out a high sound pressure level, so a dynamic mike might be in order. But in fact, many classic Miles records were recorded using a Neumann M49—hardly what one would call a conventional choice, but when it comes to microphones and miking techniques, there are no hard and fast rules.

Although engineers commonly use condenser mikes to record horns, the overtone structure of certain brass instruments sometimes is incompatible with a condenser, so instead we use ribbon mikes, such as the Beyer M160. Try one of these for a smooth solo trumpet sound. (Just don't place the microphone too close to the bell of the trumpet, as ribbon mikes are easily overdriven.)

When it comes to blazing solo sax, it's hard to beat the sound of Clarence Clemons. Dave Frazer captures Clemons' sound with an AKG C-414 placed about a foot away from the sax, midway between the bell and mouthpiece. "I chose the 414 for its presence—lots of attack and midrange," Frazer explains. "The sound was also digitally chorused and doubled. I treat his sax like a lead vocal, rather than like a section instrument."

The room plays a role in the sound of the sax. Says Frazer, "The optimum environment for recording solo sax would be a really small room with live, live, *live* walls—so live that the first reflections are nearly as loud as the sax itself." Not every

The MidiTap

Because Making Music Should Be Fun.

You've got a great studio -- the latest synths, hot software, killer sounds, cool effects, track lights.

There's just one problem. Using it is driving you nuts.

That's why we created the MidiTap. Because making music should be fun.

The MidiTap puts technology in its proper place: at the service of your creativity. It lets your equipment do its job so you can do yours: making great music.

The MidiTap is a single-space, rack-mount unit which lets you name, access, filter, merge, connect and control every piece of MIDI gear in your studio...

...**All with a user-interface that's so friendly** you'll be up and running in minutes.

The MidiTap lets you:

- Centralize control of your studio
- Enhance the capability of every MIDI device you own
- Name, configure and control your studio in your terms
- Use today's technology with ease, flexibility and power

Plus, you can create a MIDI LAN and route all MIDI data to additional MidiTaps in real time over a single, bi-directional fiber optic cable, using our MediaLink Multi-Media Communications Protocol (no more MIDI Hell!).

And that's just the beginning. The MidiTap makes live playing and jamming a joy. It's so easy to transport files from place to place that all kinds of new musical collaborations become possible.

We designed the MidiTap to grow with you. It doesn't obsolete a single piece of gear you own. Just the opposite -- it lets you benefit from your gear as never before. It catapults you into the realm of MediaLink fiber optic technology, as noted in Keyboard, EQ, Electronic Musician, Mix, Music Technology, Pro Sound News, German Keyboards and French Claviers. Plus we'll update you free via software revisions you can download in minutes with no special tools or programs.

Pros worldwide use the MidiTap live and in their studios. They love it, and we think you will, too. Because making it fun is what it's all about. Here's to your music!

The MidiTap.

Now available from Lone Wolf. Because Making Music Should Be Fun.

Lone Wolf, 1509 Aviation Boulevard, Redondo Beach, CA 90278 (800) 922-7448 (213) 379-2036 Fax (213) 374-2496

studio has such a room, so as an alternative try placing the saxophonist in a larger room near reflective, hard walls. For mellower sounds, you can record in the same place, but use a high-quality tube mike or large-diaphragm dynamic. Place the mike no closer than 1' above the bell and angle it so you can pick up sound from both the bell and mouthpiece when using a cardioid pattern. Keep in mind that sound pressure levels outside the sax bell can be well over 130db. Few mikes can handle those kinds of level without clipping or distorting.

Section Brass. The trick to miking a brass section is to retain the cohesiveness of the section sound, while maintaining control over the individual instruments. Joey Wolpert uses several different approaches to accomplish this effect. When working with musicians who are accustomed to playing together, he occasionally opts for miking the saxophones with an AKG C-24 (the stereo version of the C-12) placed 4' to 5' away from the musicians. Most of the time, however, he mikes each sax player individually, with an AKG 414 or a Neumann U87 or U47. For the trumpet players he uses a single stereo mike (or two Beyer M160s), placed 4' to 5' away from the players. When using this setup, he counts on the musicians to balance their levels among themselves.

Dave Frazer takes a different approach. For the saxes, he prefers the dynamic Sennheiser MD 441, which he feels delivers a warmer, richer sound, and creates less distinction between instruments than condenser mikes. For the trumpets, he places a U47 tube 5' to 6' back from the players. This mike also can act as a room mike for other instruments in the section. Frazer also believes in letting the musicians stand however they're comfortable and lets them find their own balances. To maintain the section sound, however, he suggests combining all the

MIKE APPLICATIONS CHART

MANUFACTURER	MODEL	TYPE	POLAR PATTERN
AKG	C 414EB	Condenser L	O, C, H, 8
AKG	C-452EB/460	Condenser S	C, O
AKG	C-34/422	Stereo condenser S/L	V, MS, XY
AKG	AKG Tube	Condenser L	V
AKG	C-747	Condenser S	H
AKG	D-12/112	Dynamic L	C
AKG	C-12	Condenser L	C
AMS	ST250	Stereo condenser S	V, MS, XY
Audio Technica	ATM25	Dynamic L	H
Audio Technica	AT4051/4049	Condenser S	C (4051), O (4049)
Beyer Dynamic	M380	Dynamic L	8
Beyer Dynamic	M160	Ribbon	H
Beyer Dynamic	M201	Dynamic S	H
Beyer Dynamic	M88	Dynamic L	H
Beyer Dynamic	MC742	Stereo condenser L	C, H, S, O, 8
Bruel & Kjær	4011	Condenser S	C
Bruel & Kjær	4012	Condenser S	C
Bruel & Kjær	4006	Condenser S	O
Countryman	Isomax II	Electret condenser S	O, C, H, 8
Crown	PCC160	Condenser S	C
Crown	PZM 30	Condenser S	O
Crown	SASS-P	Stereo condenser S	O, C
Electro-Voice	RE20/PL20	Dynamic L	C
Electro-Voice	635A	Dynamic S	O
Fostex	M22RP MS	Stereo prtd. ribbon L	MS
Fostex	M88RP	Printed ribbon	8
Fostex	M11RP	Printed ribbon L	C
Milab	VIP-50	Condenser L	V
Neumann	U47FET	Condenser L	C
Neumann	U47Tube	Condenser L	O, C
Neumann	U87	Condenser L	O, C, 8
Neumann	KM84	Condenser S	C
Neumann	RSM191	Stereo condenser S	8
Neumann	TLM170	Condenser L	H
Sanken	CU-44X	Condenser L	C
Sanken	CU-41	Condenser L	C
Schoeps	CMC 541	Condenser S	H
Schoeps	CMC 54	Condenser S	C
Sennheiser	MD 409 N	Dynamic S	C
Sennheiser	MD 421	Dynamic L	C
Sennheiser	MD 441	Dynamic L	S
Sennheiser	MKH 20 P48	Condenser S	O
Sennheiser	MKH 40 P48	Condenser S	C
Shure	SM57/58	Dynamic S	C
Shure	SM81	Condenser S	C
Shure	SM7	Dynamic L	C
Shure	VP88	Stereo condenser S	MS, XY
Sony	C48	Condenser L	O, C, 8
Tascam	PE-250	Dynamic L	C
Telefunken	ELA M 251	Condenser L	O, C, 8
Toa	KY	Condenser S	C
Toa	K4	Condenser S	C

TYPE KEY

L	Large Diaphragm
S	Small Diaphragm

POLAR PATTERN KEY

C	Cardioid
H	Hypercardioid
S	Supercardioid
O	Omni
8	Figure-8
R	Printed Ribbon
V	Variable (omni through figure-8)
MS	Mid=cardioid, Side=figure-8.

This chart is our guide to using many of the best-known mikes. If a mike not on this chart works for you, or you have applications that we don't list, you're being creative—and that's

APPLICATIONS

Vocals piano brass, general purpose, ambience
Broadcast, speech, sound reinforcement, hi-hat, rock guitar
MS & XY recording, ambience, choir, orchestra
Vocals, guitar amps
Spot-miking in orchestral recording, low-profile podium
Bass, French horn, trombone, tuba, flugelhorn, kick drum
Vocals, ambience, cello
MS & XY recording, choir, orchestra, ambience
Kick drums, timpani
Overhead, classical instruments, acoustic guitar
Kick drum, acoustic bass, tuba
Vocals, sax, trumpet
Strings, percussion, snare, acoustic guitar, hi-hat, overhead
Vocals, kick drum
MS & XY recording
Classical instruments, digital sampling, acoustic instruments
Amplified instruments
Vocals, classical instruments, sampling
Saxophone, piano, guitar
Ambience
Choir, orchestra, ambience
Choir, orchestra, ambience
Speech broadcast, kick drum, brass, sax, vocals
Speech broadcast, kick drum
Voice, piano, vibes, drums
Vocals, acoustic piano, acoustic guitar, speech
Vocals, kick drum
General purpose, vocals, overhead
Vocals, brass, hi-hat, acoustic guitar, kick drum (both pads on)
Vocals, trumpet
Vocals, piano, strings, brass, general purpose
Grand piano, general purpose, hi-hat
MS location recording, choir, orchestra, film location recording
Vocals, general purpose, percussion, alto and soprano sax
Vocals, classical instruments
Vocals, classical instruments
Film location recording, dialog and ambience
Film location recording, dialog and ambience
Toms, acoustic and electric guitar, all percussion
Vocals, speech, guitar, double bass, cello, piano, kick drum, rack toms, Leslie speaker
Piano, kick, snare, toms
General purpose, vocals
General purpose
Speech, vocals, amplified instruments, harmonica, snare, toms, electric guitar
Piano, wind instruments, acoustic guitar, violin, hi-hat, overhead
Speech, double bass, kick drum, vocals
MS & XY recording, choir, orchestra, ambience, field work
Vocals, acoustic guitar, brass
Kick drum, electric bass, vocals, amplified instruments
Vocals, ambience, classical instruments
Vocals
Vocals, drums

what it's all about

mikes to stereo, and compressing the stereo bus.

IF YOU'RE INDEPENDENTLY WEALTHY, you may want to get in on the latest craze—miking a kick drum with a Neumann U47. With a little luck, your prized 47 should last a couple of weeks. If you're feeling a bit less adventurous (or maybe just a tad cash-poor), you could use a U47FET—*with both pads on.* This results in a kick sound with a lot of attack while maintaining a tight low end—the perfect sound for up-tempo pop music and big production ballads.

Fads aside, dozens of different mikes could be called into service for recording drums. We've chosen to describe two popular methods (and a few variations).

Method #1. *Snare*: Shure SM57 or Neumann KM85 on the top, placed just outside the drum's perimeter, 2" above the rim and pointing across the drum (not directly at it) and angled toward the center. (For *brushes*, use a KM84, padded to avoid damage.) On the bottom, use a Sennheiser MD 441 or AKG 414 positioned 6" below the drum, pointed to the side of the snares and angled off the skin. Other good small mikes include the B&K 4004 and Beyer 422. *Kick*: AKG D12 or D112, or Sennheiser 421 placed about 4" in from the edge of the headless rim, 3" inside the shell, pointed below and to the left of where the beater hits. *Toms*: Sennheiser 421 or Shure SM57 diaphragm placed 2" to 3" in from the rim of the drum, pointing down and angled toward the center of the drum. *Hi-hat*: AKG 451, 452, or 460 placed several inches outside the perimeter of the hats, aimed at the spot where the drummer strikes the hat. *Overhead*: B&K "enhanced omni" 4006 or a pair of matched AKG C12s or Telefunken ELA M 251, placed outside the set looking in, about 2' above the highest cymbal. *Alternative Overhead*

ABOUT AMBIENCE

THE TONALITY OF room ambience is determined more by where you place the microphones in the room, than by what type of microphones you use. While placing mikes, remember that the farther they are from the sound source, the larger the ambient space will sound.

It's usually better to record a wide-band, full-spectrum sound that can be equalized later. Almost any large-diaphragm microphone will work, although you can use a small-diaphragm model if you just want to create an ambient sheen. But it isn't always *necessary* to create a natural-sounding ambience. You can achieve interesting effects, for example, simply by opening up the piano mikes when recording other instruments in the same room.

Here's a hot tip from San Francisco-based engineer Paul Stubblebine: When one person does multiple overdubs, place the ambient mikes in different locations for each pass. The timbral differences this creates help make the tracks sound as if they were performed by a number of different people.

—*Michael Marans*

(this setup tends to temper China cymbals, and generally doesn't require much EQ): AKG C-414s in MS stereo setup. Use cardioid for middle pattern and figure eight for side pattern. Place the mikes over the drummer's head and slightly back, high enough so the mike diaphragms can "see" every part of the kit. Bring the figure-eight signal up on two faders, and flip one fader out of phase.

Method #2. *Snare*: Shure SM57 on the top, just in from the rim, fairly close to the head, pointed at the center of the drum. *Optional*: Neumann KM84 or AKG 452 on the bottom. *Kick*: Neumann U47FET (with both pads on), just inside the shell and slightly off axis from the beater. *Hi-hat*: Neumann KM84 or AKG 452 or 460 about 1' back, angled toward the point at which the hi-hat is hit. *Rack Toms*: AKG 452 or 460 (provides tight low end and lots of attack). *Floor Tom*: for the top, Neumann U87 placed 6" above the head, pointing toward the center; for the bottom, a Shure SM57 (out of phase), panned to the opposite side of the stereo image. *Overhead*: Two AKG C-12s in XY stereo, 4' above the cymbals. In this setup, most of the sound comes from the C-12s; the individual mikes are used to fill in the sound.

If you're recording an R&B song, try dynamic mikes on the kick (Sennheiser 421), snare (Sennheiser 409), and toms (Sennheiser 412). The resulting sound will be solid and punchy. •

If A•DAM Is Just Another Digital Multi-Track, Stevie Is Just Another Musician.

You know how good this guy is. For total musical abilities there's simply no one better. So when an impressive guy like Stevie tells us he's real impressed with A•DAM, its music to our ears.

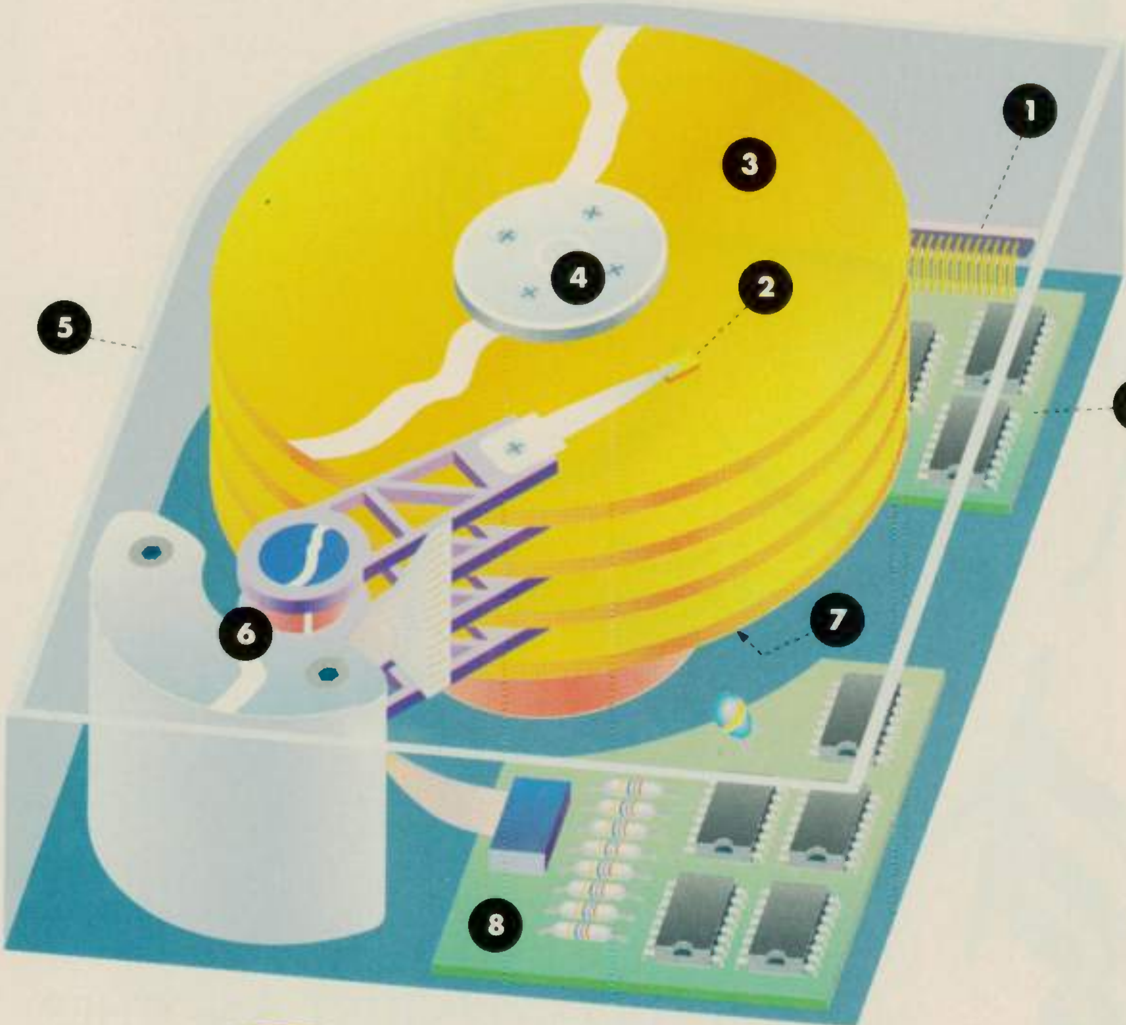
Well, what Stevie is doing for music, Akai is doing for digital multi-track. A•DAM. The world's first portable digital multi-track, with the convenience of universally accepted 8mm tape, gives you the flexibility to expand to 36 tracks. It's the only digital machine within the means of any professional application.

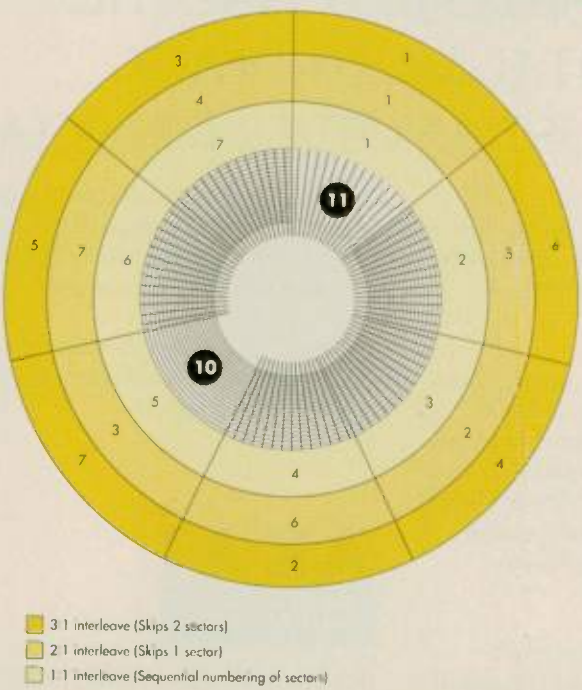
A•DAM is the ultimate satellite multi-track. In a midi facility, A•DAM is the ideal primary machine because its SMPTE compatibility lets you devote 12 digital tracks to vocals. And with single frame editing, you can now eliminate the most difficult breaths and pops with absolute precision.

A•DAM. Just another wonder from Akai. Call today for the authorized AKAI Digital dealer nearest you.

AKAI
DIGITAL

P.O. Box 2344 • Fort Worth, TX 76113
(817) 336-5114





Hard Disk Drives

By Bryan Lanser

Illustration by Rick Eberly

IMAGINE FLYING AT MACH 2 in a 747, 15 feet off the ground. . . This is the equivalent of what it is like to be the head of a Winchester disk drive flying over the surface of a magnetic platter. This considered, it's hard to believe that hard disk technology exists at all, much less that it has matured at such a remarkable rate over the last ten years. The standard drive in the early '80s was typically 5 or 10 *megabytes* in a full-height package with average seek times greater than 30 milliseconds. Now we are on the brink of having 1.2 *gigabytes* in the same sized package with seek times close to 10 milliseconds.

Indeed, Winchester disk drives have come a long way since their early days in IBM and DEC mainframes. It is interesting to note, however, that the basic components all have remained pretty much intact from the early days. Let's take a look at How It Works:

1. Drive Interface. Until recently, the most common drive interface was the ST-506 used in early MS-DOS computers. The need for faster interfaces led to the development of the ESDI (pronounced "ezdee") interface, and recently the SCSI (pronounced "scuzzy") interface has become very common.

2. Heads. These are very small, light record and playback heads that are aerodynamic in design so they "fly" over the surface of the disk. Their gaps are quite thin since data is being read from or written to the disk at extremely high rates.

3. Platter. The platter is the rotating recording media in a hard drive. It usually is composed of an aluminum substrate coated with a magnetic emulsion, similar to audio recording tape. Some drives use a plated media that is less susceptible to surface damage in the event of impact to the drive mechanism. Drives can have from one to nine platters.

4. Spindle. High-speed motor and shaft assembly that spins platters at a fixed rate (constant angular velocity). Disk drive motors typically run at 12V or 24V.

5. Disk Chamber. This modular unit houses the platters, spindle, heads, and actuator. Its most important function is to keep dust away from the heads and platters.

6. Actuator. This head-positioning mechanism consists of a motor that positions the head, and some kind of mechanical linkage to convert the rotary motion of the motor into the linear motion of the heads. Less expensive drives use stepper motors, which are less accurate and slower, therefore increasing the amount of time necessary to seek out a specific track. Since stepper motors have inherent phase-to-phase variation and a great deal of mechanical inertia, they limit how thin a track can be and thus limit how much data can fit on a platter. Larger capacity drives use *voice coil actuators*, a refinement of the technology used in driving a speaker cone. This technology can provide extremely fast and precise movement of the head assemblies, thus allowing for thinner tracks, greater density, and faster seek times.

7. Servo Platter. Traditionally, this is the bottom-most platter. It contains the head positioning information that is used to locate a specific track.

8. Read/Write Amp. This amp is optimized for noise-free amplification of disk data during a read operation, and it generates the current necessary to write Data onto disk. Better quality drives place the amps on the head arm to minimize the distance that the minute head signals must travel before being increased in level.

9. Controller. The controller is responsible for all drive operations: organizing the data on the disk, writing or reading data, and telling the head positioning mechanism where to go. It also serves as traffic manager to organize the flow of data from the drive to computer and vice versa.

10. Tracks. These concentric rings (also known as cylinders) contain the actual digital data, written as alternating N-S (north-south) orientations of the magnetic domains of the magnetic material coated on the platter.

11. Sectors. Each track is divided into sectors, typically 512 bytes long. These sectors serve as road markers for the drive controller to locate the positions of specific data. Without sectors, the data could not be retrieved or stored accurately.

SEVERAL OTHER IMPORTANT ASPECTS of hard drives can't readily be depicted graphically:

Directory. Also known as the file allocation table, this is the table of contents on the disk, used to locate specific files. Each time a file is created, modified, or deleted, the directory is updated to reflect the changing data on the disk. When the directory is full, the drive knows that no more data can be written in any of the sectors on the drive surfaces.

Partitions. Subdivisions of the disk space, are provided primarily to help the user organize the data storage space in a more efficient manner. They can be "hard" (actually written into the data on the disk) or "soft" (imaginary divisions of the data into smaller chunks).

Data Encoding Technique. Just as there are different ways of encoding audio data on tape (analog versus digital, for example), there are different ways of encoding data on a disk drive platter. One technique called MFM (Modified Frequency Modulation) is common in drives using the ST-506 interface. Larger and faster drives use the RLL (Run Length Limited) encoding technique. RLL data encoding techniques offer 50% more data density than MFM techniques.

Actuator Lock & Landing Zone. The actuator lock keeps the drive heads in a fixed place, called the "landing zone." If the drive is moved and the heads bounce against the surface, only the magnetic material in the landing zone will be damaged. No data is stored in this zone.

Standard Recording. Since the platter rotates with constant angular velocity, and the linear track length increases as you move farther away from the center of the disk, the linear data density is highest at the innermost tracks (about 20,000 bits/inch). As the track length increases, the data density decreases to about 12,000 bits/inch at the outermost tracks. Although this technique is the standard, it wastes much of the available recording area toward the outer tracks of the disk.

Zone Bit Recording. ZBR is a technique of maximizing the data density on the disk by making it constant (usually 20,000 bits/inch) regardless of the track position. Since drives exhibit constant angular velocity (fixed rotational speed), constant linear data density causes much higher data rates at the outer tracks; hence the data transfer rates of ZBR drives vary between 9 to 15 megabits/second from innermost to outermost tracks. ZBR drives use hash marks instead of hard radial sectors.

Interleave. Some computers are not capable of assimilating data as fast as it comes off a hard drive. Without using any tricks, this would require reading or writing a track/sector, then waiting for another revolution to access the next contiguous track/sector. Interleaving is a way to read and write tracks in every second or third sector in order to minimize this delay. A 2:1 interleave skips every other sector; a 3:1 interleave reads and writes every third sector. •

Thanks to Jerry Burgiss at Storage Dimensions and Bart Asquith at Seagate for their assistance.

THE BIG PICTURE

Visual Thinking

BY ROBERT WAIT

"Musicians usually make good filmmakers because they understand pacing and rhythm."

THE FIRST TIME I HEARD THAT statement, I was teetering on the brink of a film career after spending most of my formative years playing music. I suddenly had realized that two of my favorite things, music and film, were so harmonious that I could combine them into a great career. Ten years later, I'm still elated whenever I create or see a combination of film, audio, and music that moves the viewer emotionally far beyond where either of those elements could on its own.

Yet a successful combination doesn't come easily. Writers write differently for film than for novels, and composers compose differently for film than for non-film projects. It all comes down to one basic concept that's a bit difficult to define, but remains essential to the creation of well-made films and videos. Let's call this concept *visual thinking.*

Are You Composing In The Key Of See?

Visual thinking means telling a story with pictures rather than words. Whether you're working in audio or scoring, or some aspect of production, by understanding the visual rhythms that writers, directors, and editors try to set up when making a film, you'll be much more in sync when it's time to add your contribution.

Visual thought begins with the screenwriter. A sentence such as "John is angry and frustrated" has little place in a screenplay because it *tells* rather than *shows* how John feels. A visual interpretation would be "John smashes the window and curses the passing mockingbirds for ruining his life." By seeing how John reacts to his emotions, we know how he feels.

Much as a composer uses the orchestra's different sections to convey the dynamics of a symphony, a screenwriter controls the script's dynamics by describing characters' actions and reactions surrounding a central conflict. Drama is fueled by conflict, and a story without a problem to overcome lies flat, with no forward momentum. A composer's job often entails writing music that "fixes" scenes which have lost their momentum or picked up too much. No matter how much effort is put into writing, directing, and editing a film to achieve perfect pacing and dynamics, certain scenes that are essential for plot purposes will fail dramatically somehow, and will need some help from the composer.

Making A Scene. Generally, a film can be broken down into beginning, middle, and end. Each part, or "act," can be broken down further into beginning, middle, and end. Each act contains many scenes (each with its own beginning, middle, and end) that pit the main characters against additional mini-conflicts that they must overcome in order to move forward to the resolution of the main conflict.

When piecing together a film, the director and editor cut individual images into scenes, then assemble the scenes in their most effective order. The composer and sound editors then are brought in to interpret the dynamics of each scene and the way it relates to the overall picture. This is why soundtrack music often doesn't sound nearly as interesting without the imagery it accompanies: Unless music is used to alter the pace or mood of a non-working scene, the music's primary responsibility is to go unnoticed while it adds punch and nuance to the visual dynamics. (If the visual scene is extremely big and exciting, the score can get away with being "big," as was the case in such films as *Lawrence of Arabia, Out Of Africa,* and *Raiders Of The Lost Ark,* or in climactic scenes such as the night run to the cave in *Dead Poets Society.*)

The sound editor's goal in film is to duplicate and enhance the visuals so effectively that, even with eyes closed, the viewer might fully imagine the scene simply by hearing the soundtrack. Visual imagery in film shows you the "stage" where the actors are performing their parts, but "visual audio" brings that stage to life.

Exercising Your Visuals. Blades of grass wave in the breeze. Light footsteps rapidly approach. A flash of brown fur brushes against a bush, which softly rustles in response. Four legs, two eyes—frightened, intent, unblinking. The legs leave the ground, taking the animal over the fence and onto greener grass on the other side. In a flash he's gone, as the sharp barks of distant hunting dogs fade away down the wrong path somewhere in the soundscape. A fence brace nicked by the fox's leg gives up its hold and falls to the dirt with a thud. All is quiet again.

Verbal interpretation: *A quick brown fox jumps over a lazy fence.*

If you think about it, nearly every situation can be imagined as a montage of visual and aural images. If you're involved in audio for picture, exercise visually on a regular basis. As you see everyday events take place in your environment, add your own score and sound effects. This type of visualization will help tremendously in your work. •

Robert Wait is a post-production supervisor, film director, editor, and composer working in Hollywood.

Digital Interface Formats

By Bryan Lanser & Jeff Burger

THE ADVENT OF THE digital audio recorder brought with it the need to transfer digital audio data between machines.

Unfortunately, manufacturers implemented several incompatible protocols for the transfer (encoding/decoding, or "transcoding") of digital data.

This chart will help you determine which products perform the conversions you need.

DESTINATION

SOURCE	AES/EBU	SDIF-2	S/PDIF	PD	Yamaha	Akai
AES/EBU		B,C,D, E,F,L,M,Q	A,C,D,E, F,G,L,M,Q	C,D,E, F,I,L,M,Q	E,K,L,M	N
SDIF-2	B,C,D,E, F,L,M,Q		C,D,E, F,L,M,Q	C,D,E,F, H,L,M,Q	E,L,M	O
S/PDIF	A,C,D,E,F, G,L,M,P,Q	A,B,C,D, E,F,L,M,Q		C,D,E, F,L,M,Q	E,L,M	
PD	C,D,E, F,I,L,M,Q	C,D,E,F, H,L,M,Q	C,D,E, F,I,L,M,Q		E,L,M	O
Yamaha	E,J,L,M	E,J,L,M	E,J,L,M	E,L,M		O
Akai	N	O		O	O	

CONVERSION KEY:

- **A.** Audio Design Pro Box 1: transcodes S/PDIF with AES/EBU.
- **B.** Audio Design Pro Box 2: transcodes AES/EBU with SDIF-2.
- **C.** Harmonia Mundi Musica: transcodes AES/EBU, SDIF-2, S/PDIF, and PD (DUB C); sample-rate conversion.
- **D.** WaveFrame AudioFrame: transcodes AES/EBU, SDIF-2, S/PDIF, and PD (DUB C); real-time sample-rate conversion.
- **E.** Studer Editech Dyaxis:transcodes AES/EBU, PD (DUB C), SDIF-2, Yamaha, and S/PDIF.
- **F.** NED Direct-To-Disk: transcodes AES/EBU, SDIF-2, S/PDIF, and PD (DUB C).
- **G.** Digidesign Sound Tools: transcodes AES/EBU and S/PDIF.
- **H.** Otari CB-503: transcodes PD (DUB A/B) and SDIF-2.
- **I.** Otari CB-505: transcodes AES/EBU with PD (DUB A/B).
- **J.** Yamaha FMC1: converts Yamaha 2-channel format to SDIF-2, S/PDIF, and AES/EBU.
- **K.** Yamaha D-2IF: converts four 2-channel pairs of AES/EBU to 8 channels of Yamaha format.
- **L.** Yamaha DMP7D: accepts Yamaha multitrack, SDIF-2, PD multitrack (DUB A/B), and two channels of AES/EBU or S/PDIF. Outputs two channels simultaneously in S/PDIF, AES/EBU,Yamaha DIN, as well as SDIF-2 or two channels of PD multitrack (DUB A/B).
- **M.** DAR DASS 100 & Soundstation II: transcodes AES/EBU, S/PDIF, SDIF-2, PD (DUB A/B), and Yamaha 2-channel formats. Sample-rate conversion, (DASS 100 only).
- **N.** Akai DIF1200: transcodes Akai with AES/EBU with selection of which two channels will be addressed.
- **O.** Klotz AFC-1.12: simultaneously transcodes 12 channels of Akai format with SDIF-2, PD (DUB A), and Yamaha multtrack formats.
- **P.** A cable that unbalances this signal will work, providing that level and clock differences are not a problem.
- **Q.** AMS AudioFile: transcodes SDIF-2, S/PDIF, PD, AES/EBU.

Digital Audio Interface Formats

AES/EBU: Developed by the Audio Engineering Society in tandem with the European Broadcasting Union, this 2-channel digital data interface is found on most professional digital audio systems. Interconnection is balanced, RS-422 using XLRs (although some manufacturers use D-sub connectors).

SDIF-2: The Sony Digital Interface Format was developed by Sony for videotape-based PCM processors and digital open-reel multitracks. Interconnection is via unbalanced 75Ω BNC lines for processors and via balanced D-sub connectors for multitrack machines.

S/PDIF: The Sony/Philips Digital Interface Format is used primarily in consumer digital audio equipment such as CD players, DAT machines, and F1-type processors fitted with digital adapters. S/PDIF has the basic attributes of the AES/EBU protocol but in unbalanced form. Interconnection is via unbalanced 75Ω RCA jacks or fiber optic lines.

MADI: The Multichannel Audio Digital Interface was co-developed in England by Sony, Mitsubishi, Neve, and SSL. It has a maximum capacity of 56 channels of 24-bit audio and features AES/EBU-compatibility in 2-track format. Interconnection is via unbalanced BNC coaxial link or an as-yet-unspecified fiber optic link. MADI requires single input and output lines. We know of no MADI conversion products at this time.

PD: The Professional Digital format (also called ProDigi) was developed by Mitsubishi and later adopted by Otari for use in their open-reel digital multitracks. Interconnection is via one of two different D-sub connectors: DUB connectors handle 16 channels each of a 32-channel signal; DUB C is used when performing digital dubbing to or from a 2-channel device.

Yamaha: The Yamaha format is present on the rear panels of some Yamaha products. It is essentially a stripped-down version of S/PDIF and AES/EBU with two variations: One handles 2-channel communications via an 8-pin DIN, while the other handles multitrack formats through a 25-pin D-sub connector.

Akai: The Akai protocol is implemented on the A-DAM DR1200 transport and supports 12 channels simultaneously via a 37-pin D-sub. •

GUEST EDITORIAL

◂◂ PAGE 10

gives a share of the ASCAP or BMI residuals to the ghostwriter as compensation for not taking any credit on a project.

• Can you claim credit for working for the composer? If this is a problem for the person for whom you're ghosting, ask for the right to use that person's name on your resume without revealing specific information about work done for the employer. This will protect employers from letting it be known that they used a ghostwriter on a project. At the same time it will entitle the ghostwriter to claim valuable experience. A composer who starts ghosting for another composer must assume that he or she will receive the least amount of recognition possible. Then, if the "big" composer decides to credit the ghostwriter, it's icing on the cake.

• Finally, ask yourself this: Is monetary compensation alone (without any credit) sufficient?

If the established writers who use outside composers would remember what it was like to struggle at the start of their careers, maybe more of them would treat the people helping them with some added respect. It can be extremely demeaning for a ghostwriter to be expected to create music for someone else, be paid relatively little money, and get little or no credit out of the deal.

If more people in the business would take a firmer position and stand up for their rights as creative forces, the more established composers would have a smaller pool of low-budget writers to draw from when they need help finishing their high-budget work. •

SHARP ANGLE

◂◂ PAGE 12

cation, collide with the time required to develop musical skills and knowledge of theory. If players now select a sound from a factory-prepared palette, will future players merely assemble riffs pre-digested by algorithmic composition programs?

The trivialization of popular music erodes the foundation upon which its popularity is based (its ability to strike a chord in the emotions of the listener). If listeners no longer listen, music is no longer essential, and no matter how many dBs of signal-to-noise ratio can be achieved by the latest in techno-gadgetry, the recording industry will be in trouble. •

CRAIG DORY

◂◂ PAGE 20

Setting up in that church was a nightmare. But the recordings came out great.

Boston's Symphony Hall.) Built above a savings bank in 1875, the Troy Savings Bank Music Hall has been host to everybody from Fritz Reiner to Yo Yo Ma to Katherine Battle. It is smaller than most halls Americans are used to—it seats 1,200—but offers considerable acoustic advantages because it is a true concert hall, lacking proscenium and fly space, and not just a modified theater.

Within this space Dory has installed a remarkably simple yet powerful digital recording system, containing almost no stock equipment. The mikes are modified Neumanns, B&Ks, and Schoeps. "We've had [California-based microphone expert] Stephen Paul take out the 6-micron diaphragms in some of the mikes and replace them with 1.5-micron diaphragms. It's not cheap, but oh boy. We also modify the electronics. Phantom powering is an abomination; it puts more circuitry between you and your signal than what you want. So we separate signal and power leads, and clean up all the components in general. Schoeps mikes, for example, have a built-in 3dB rolloff starting at 25Hz. I don't want that, so I changed a capacitor and the rolloff is no longer there. You wouldn't believe what that does for group delay."

The audio cable is custom-built for Dorian Recordings by a supplier in Colorado. It is 100% copper tube-shielded, and its characteristic impedance is matched on a *per-cable* basis to the source and target impedances of the equipment to which it connects. Cables never run more than four meters, and usually less, terminating in two-stage record electronics at the base of the microphone stands. The first stage is a not-quite-stock, twin-servo, Jensen mike preamp (with some component changes worked out by Dory and the manufacturer). The second stage is a special A/D converter. Originally built under the auspices of Apogee Electronics, it currently is on the fourth of an endless series of in-house redesigns. Once converted

PRODUCER'S TIP

"After each session we take copious notes as to mike placement and details of the recording, and the next time we go into a similar situation, we start from that point. If we've done a piano quartet or trio, or solo piano, the next time we do any of those, we go back to the files and use what we started with the last time—even though we will have to change things because the repertoire will be different; recording Mozart is a hell of a lot different from recording Shostakovich. But we will at least start with where we left off, and continue working from that point."

—Craig Dory

from analog to digital, the signal travels by fiber-optic cable to the control room, where it goes straight onto DAT, with monitoring provided by a Wadia D/A converter, Boulder Electronics amplifier, and audiophile Nestorvic speakers.

Dory hasn't made up his mind yet about his favorite DAT deck. "We have seven professional DAT decks, all from different manufacturers. The difference between them is the transports, and we're still trying to find a transport we really like. We're dragging our feet because we are waiting for [a deck equipped with a standard for] SMPTE time code. I think the machines at that level will be a little more serious." Master editing is performed with the Sonic Solutions system, and mastering and pressing is handled by JVC's plant in Tuscaloosa, Alabama.

Not all classical recordings can be made under the tightly controlled conditions available in the Music Hall, however. In the field, the Dorian trait of Obsessive Attention To Detail can present quite a challenge for Dory and his crew of four full-time engineers.

"On our most recent trip to Europe," he recalls, "We recorded the Great Organ of St. Eustache, in the Les Halles section of Paris. It was the most horrific experience in my entire life of recording. Les Halles used to be a market, 20 years ago, where all the farmers would sell their produce. The market has moved outside of town and Les Halles is now a big park, but under it is a major Parisian subway station—like 42nd Street in New York, except bigger. So we had subway noise to deal with. We had to fly the mikes from the gallery level, run the fiber-optic cable, convert a stone room into an acceptable control room—it was just a little live, shall we say—and deal with the fact that while the weather was a humid 90 degrees during the day, every afternoon there would be a thunderstorm and the temperature would drop 20 degrees, playing havoc with the tuning of the 8,000 pipes in the organ.

"But that wasn't the worst. We were recording two days prior to the official celebration of France's bicentennial, and they were going to be shooting fireworks off the top of the church we were recording in! This is a 600-year-old church, and the French are shooting fireworks off the top of it? Interesting, to say the least. I had a 450-foot, fiber-optic link that went from the back of the church, where the mikes were, to our control room in an adjacent building. It actually wrapped around gargoyles on the outside of the church. Well, one of the fireworks technicians saw our cable, which wasn't his, and so he gave it a yank—which ripped it out of its socket, destroying it. I had to send back to the States for an overnight replacement. Setting up in that church was a nightmare. But the recordings came out great." •

For a catalog of Dorian CD releases, contact Dorian Recordings, 17 State Street, Suite 2E, Troy, NY 12180.

IT'S ABOUT TIME

◄◄ PAGE 34

sounds and two rests.

When the Emax's CRUZ function is on, holding down two notes causes the note value to double; holding down three notes causes the rhythm to triple, and so on. If you set the basic note value to quarters, for example, you can alter the rhythm in real time by pressing (or releasing) additional keys.

Adding the CRUZ feature to the sounds in Fig. 3 would produce some interesting patterns. For instance, holding down both the *C* and the *D* keys produces a fairly busy eighth-note pattern. Since the major thirds stacked above the *C#* are all empty, holding down the *C* with the *C#* together creates a more moderate eighth-note groove. Holding down the *C*, *C#*, and *D* produces triplet patterns, and adding the *D#* produces sixteenth-note rhythms.

Since most current arpeggiators can slave to external sync or send their own MIDI timing information, it's easy to interface an arpeggiator to a software sequencer. If you record your arpeggiated masterpieces in the sequencer, you can tweak them even further. Have fun! •

Euphonix Crescendo Digitally Controlled Console

AT FIRST GLANCE, THE Euphonix Crescendo looks like something other than a major-league recording board. Where is the familiar vast array of knobs and switches? Where is the huge console frame? If you're looking for a huge, big-name console to impress the naive by appearance alone, the Crescendo may disappoint you.

If, however, you're looking for a professional, surprisingly powerful multitrack mixer that offers complete automation of *every* function, the Crescendo may be the one. Its automated design is not completely revolutionary: It follows in the footsteps of the Harrison Series X and Trident DiAn. What is revolutionary, however, is the Crescendo's price. While the Harrison and Trident sell for $400,000 and more, the Crescendo's price is a fraction of that. This alone deserves notice. Even if you don't plan a purchase, you should be familiar with this board: The Crescendo surely is a portent of consoles to come from other builders.

With this much promise, does the Crescendo deliver?

OVERVIEW

The Crescendo is a digitally controlled, analog mixing console. All audio signals remain in the analog domain. Routing, levels, EQ settings—in short, everything—can be programmed, recalled, and automated digitally. In tandem with some unconventional approaches to signal routing, this results in a mixer that demands an open mind on the user's part—not only to make the most of the board, but also to understand its basic concepts.

For instance, we tested a "Crescendo 5624," which is a 56-fader/28-channel, 24-group board. Yet even that description is deceiving. Because of the Crescendo's unique design, it's not possible to describe it in standard terms (such as 56 inputs x 24 groups x 28 tape monitors). Its 28 channels sport a total of 56 microphone inputs and 112 line inputs, routable through 56 faders. Any fader can be mono or stereo—or receive up to four line plus two microphone sources *simultaneously*. Any input can serve as tape monitor. Any input fader can be programmed to be a group master fader.

This is not your father's Oldsmobile.

SYSTEM COMPONENTS

The Crescendo consists of two major components: a *mix controller* and an *audio mainframe*. From a distance, the controller looks like an 8- or 16-track mixer. It houses the system's knobs, faders, and switches, and several internal microcomputers.

The mainframe connects to the controller via a data link cable (up to 50' long). Behind the mainframe's peel-away fabric cover are all the audio cards. For the most part, audio control is performed by DCAs—*digitally controlled amplifiers*. Unlike their voltage-controlled counterparts, VCAs, DCAs are capable of perfectly accurate level control and very low distortion. As we'll see, however, in some cases DCAs suffer from unwant

PHOTOS: PAUL HAGGARD

IN REVIEW

ed sonic side-effects.

All audio connections are made at the mainframe; no audio signals actually pass through the controller. Instead, the controller serves as a combination user-interface and remote control, not unlike the way a Lexicon 480L LARC acts as both interface and remote control for its own blank-faced, rackmount digital processor.

Euphonix also offers a third component, a stand-alone 386-type IBM PC (or clone) computer, which is not required for console operation or audio processing. This "host computer" delivers Euphonix' Mac-like MixView graphic interface. The computer also serves as an archival system for console settings, allowing floppy or hard disk storage. It connects to the controller via RS-232 ports.

Finally, a custom rackmount patchbay is available from Audio Accessories of Marlow, New Hampshire. Our test unit has a 576-point TT (bantam) bay. Tape decks, outboard effects, and other source/destinations are connected to a central ADC punchblock wired to the patchbay. Multipair audio cables run between the audio mainframe and patchbay; both ends terminate in multipin Elco connectors.

Inside The Mix Controller. The controller is small: just over 7" thick (including the integral LED meter bridge), 30" deep, and 53" wide. It's the right size for a single operator, cozy for two people, and a tangled game of Twister for three.

The controller has a bolted steel chassis that mounts on a steel stand. You can put your full body weight on the center of the controller with no noticeable stress on the chassis, and a corner can be lifted with no noticeable warping. The ends are capped with ABS plastic, the same stuff used in whitewater canoes and bicycle helmets. Sadly, gone is any trace of a plush, tuck & roll-padded armrest. We're left instead with an ABS armrest, mottled to feel and look like splatter-painted metal. Some will grumble about this aesthetic point, but remember, it takes an open mind to appreciate the Crescendo.

Poking through the controller's top surface is a moderate number of switches, knobs, and faders. The surface material is a plastic-like Lexan covering, finished in light and dark shades of grey. While some markings and labels (such as "source" or "solo") are screened in white, many indications are translucent, visible only when backlighted by LED. It's difficult to read the surface in dim light, due to poor contrast between the grey surface and the markings. Euphonix plans to redesign this.

The Euphonix-designed faders are *très cool*. They are rubber-coated and asymmetrical; the far end of the fader is higher than the end closest to you. This lets fingers drop into them most sensuously (especially if you're into rubber). This more than makes up for the missing padded armrest.

The switches deliver a satisfying tactile "click" when depressed, and the rubber-knob pots feel smooth. (Remember, all switches, pots, and faders are digital controllers.) In subdued light it's almost impossible to tell that the knobs' pointers are color-coded. The controller's master section includes two other interfaces: an easy-to-read alphanumeric display, and a data entry wheel, similar to jog wheels found on video editing gear.

All controls are laid out cleanly and logically. Everything is within easy reach, and feels right. The mix controller is attractive, though I'd forsake the corporate grey for a more flamboyant use of color coding, and more legible markings.

Open The Pod Bay Door, HAL. Our test model had a total of nine "pod bays," or module slots, holding seven input/output (I/O) control modules and one master control module, plus one blank panel. Currently, Euphonix can offer up to 15 slots. To operate, the controller needs at least one I/O module and one master module.

The I/O modules each control four *I/O channels* and associated aux sends, trims, and so on. (As we'll see, each I/O channel has two faders, and in most respects can function like two independent stereo channels.) The master module controls a *master section*—with monitor controls, master aux sends, two stereo output faders, etc.—and the *command section*, through which all programming and automation occur. Each module measures about 5.5" wide, and runs the controller's 30" depth. The modules angle upward to hold their associated sections of the meter bridge.

Removing modules from the board is a dream: Just power down the controller and lift off the panel above the meters, and you'll expose the handles for the modules. Grab a handle, pull up, and you'll have the entire module in your hand; no screws, no muss, no fuss. You'll see that each module is a heavy, enclosed steel box. The only electrical-like thing about it is a gold-plated, multipin male connector, recessed through a slot on the bottom panel, which connects to an associated female connector on the bottom of the controller chassis. Inside each module box is a glass-

EQ Lab Test

Product: Crescendo
Manufacturer: Euphonix, 441 Page Mill Rd., Palo Alto, CA 94306, (415) 325-5003
Price: ~$95,000 (Crescendo 5624); ~$80,000 (Crescendo 4824). Other prices upon request. Prices include remote patchbay, cabling, 386 PC host computer, MixView software.
Frequency Response (line in to mix out):
Claimed: 15Hz - 30kHz (+0, -0.25dB).
Tested: 10Hz - 35kHz (+0, -0.25dB); 10Hz - 110kHz (+0, -3.0dB).
Signal-To-Noise (OVU, 22Hz - 22kHz bandwidth, unweighted):
Claimed (line to mix): 90dB.
Tested: 90.5dB.
Claimed (24 lines to mix): 80dB.
Tested: 80.5dB.
Microphone Input Noise (EIN):
Claimed: -126dB
Tested: -126.5dB
Distortion (@1kHz, unweighted):
Claimed: < 0.005%THD and IMD
Tested: 0.002%THD; 0.004%IMD
Maximum Input Level (@1kHz, rated distortion):
Claimed (mike to mix): +28dBu.
Tested: +29.5dBu.
Maximum Output Level (@1kHz, rated distortion):
Claimed: +28dBu.
Tested: +27.5dBu
Crosstalk (between mix buses):
Claimed (@ 1kHz): -90dB
Tested: -94dB
Claimed (@ 10kHz):-80dB
Tested: -82dB

epoxy motherboard containing the digital control circuitry, ports for future boards, and an internal microcomputer—one for each module. (The master module holds two microcomputers: one for the master section, and one for the command section.) Soldering is impeccable; components are well laid-out and labelled.

To reinstall the module, you place its front edge into a slot near the armrest, and lower it into its snug home. No way can the connector miss the mark: The machining is perhaps the finest-tolerance metal work I've seen in a console, regardless of price. An added plus lies in the fact that there are no exposed circuit boards. Well done.

The Monolith. The audio mainframe stands about 50" high (as an option, it can be rackmounted). At the bottom is the power supply and master audio board. Farther up are the audio boards—up to 28 in all. (Each board contains the audio electronics for one I/O channel.) You can slide the boards in and out of the frame's slots by moving two thumb-locks. Audio cabling in the mainframe appears well-shielded.

The console is designed for +4dBm balanced operation. If a -10dBV multitrack were to be patched in for mixdown, however, we found enough headroom in the microphone preamps for them to function as -10 line inputs.

The front of the black mainframe is covered by a fabric-covered grill that pulls off to reveal access to boards and slots. On the back, mating to Elco multipin connectors, are several multipair audio cables that snake their way to the patchbay. The data link cable port also is on the back; the actual link is opto-isolated, which electrically isolates the controller and mainframe from one another. For safety's sake, and to minimize ground loop potential, a large ground lug is on the back of the mainframe, to which everything in the studio can be "star grounded." The mainframe's 1.75kW power supply requires 120VAC, 15 amps.

The power supply's fans are a bit noisy, and you can hear the metallic clicks of electronic relays during system operation. While you could track with an open mike in the same room as the mainframe, we recommend installing the mainframe in a separate machine room.

OPERATING THE CRESCENDO

In the traditional mixing console, most knobs, switches, and faders perform only one function apiece: An EQ IN/OUT switch does that, and nothing else. While this leads to a lot of hardware, it also results in easy, fast operation; once you understand the meaning of an IN/OUT switch, you can operate one on any console.

Euphonix has given us some dedicated controls on the Crescendo, such as ON/OFF switches (for input channels, stereo outputs, aux sends, etc.) and a STEREO OUTPUT fader. But most are "soft"; they perform multiple functions. It would be exhausting to cover all the steps in each programming option, so we'll look at three main approaches.

(Some of the interface software for the stand-alone PC host computer, by the way, is still in development. Our test model provided HELP SCREENS, graphic display of EQ curves, a FILE TRANSFER screen, and more.)

"Dedicated Button" Options. Programming the Euphonix is accomplished by "calling attention" to a given section, then choosing your options. An example: There are three independent monitor sends (MON A, B, and C) to feed control room monitors or headphone amps. Next to each level control is a button marked SET. Beneath the controls is a bank of 12 dedicated MONITOR SET buttons: LEFT, RIGHT, STEREO, MONO, AUX 1-2, 3-4, 5-6, 7-8, STEREO (OUTPUT) 1, ST2, EXT1, and EXT2. Any monitor send can get its signal from any or all of these 12 sources.

To assign the sources MON A receives, you first "call attention" to the section by pressing the SET button, then press your choice of sources. That's it. As soon as you move on to MON B's SET button (or any other "attention-type" button), you lock in your choices for MON A, until you reprogram, or until you call up another preset "snapshot."

Monitor assignment is one of the easiest things to program on the Crescendo, since you have a bank of dedicated buttons from which to choose. Two other main types of soft-key programming require more thought.

Numerical Pad Options. The second approach requires using the numeric keypad and several "soft" buttons located underneath the alphanumeric display (in the master module's command section). Let's say you want to assign upper input fader 17 to group output 21. First you call attention to the input fader by pressing its BLOCK ATTENTION KEY beneath its associated ASSIGN indicators. The display will say:

17UF—> MODE SRC ASGN

The first entry indicates "upper fader 17" and "—>" indicates "block attention" for that fader. Beneath "17UF" and the three other names is a button. Since you want to assign a group, you press the button underneath ASGN. Then move to the keypad and type "21." Next, type "." to enter the selection. You can assign multiple groups (including direct and stereo outputs) by typing in their numbers, followed by a decimal point. This is trickier than using dedicated buttons, but logical enough.

Data Entry Wheel Options. The third soft-key programming approach is for more complicated procedures, such as setting EQ values or choosing modes for an aux send or input fader. Similar to the last approach, it uses the data entry wheel instead of the keypad to scroll through options (such as aux send or fader modes; see "Auxiliary Sends" below).

The Interface: Obliquely Intuitive. The Crescendo offers a couple of other option-select variations. In some modes, the host PC computer and mouse can be used to select options displayed on the computer screen. In other modes, two buttons located near the data wheel serve as increment/decrement (+/-) buttons, letting you step through options one by one, instead of spinning the data wheel. Unfortunately, these buttons also do double-duty as ASGN and SOURCE buttons, so if you're not in the correct edit mode, you may accidentally select a source, rather than step through options. Dedicated data-step buttons would be better.

Copying the settings from one section or I/O channel to another is performed easily in a few keystrokes. This is a real time-saver when setting up multiple stereo channels or similar mono channels.

Like most software programs, some rote learning is required. It took an hour's worth of reading and an hour's worth of use before "*Now* I get it!" bells began to ring in my head. Some functions are intuitive, others are oblique. What I like least about the Crescendo's interface is the lack of consistency in assigning buttons and choosing options: Sometimes you use dedicated option buttons, sometimes the keypad, sometimes the data wheel.

The difficulties inherent in using software often are outweighed by the advantages. When I started to "get it," it became easy to use the board, despite my years of reaching for dedicated knobs and switches. In fact, it is easier to master the Crescendo than it is to program most synthesizers. So, while there's room for software refinements, I like the Crescendo's basic interface. It's obvious that much thought has gone into its development.

SIGNAL ARCHITECTURE

When it comes to programmable routing options, I know of no other console as flexible as the Crescendo. The chart below compares its I/O channel with the typical I/O channel found on a conventional board.

Now, consider this: On a conventional board, you're limited to a mike or line source for the long fader, and a line or tape source for the short fader; aux sends usually receive whatever is sent to the long fader; only a few boards let you "split" the EQ so that you can add minimal EQ to the tape monitor while you equalize the main channel. None of these limitations apply

CONVENTIONAL I/O	CRESCENDO I/O
• 1 mike input	• 2 mike inputs
• 1 line/tape input	• 4 line/tape inputs
• 1 long-throw mono fader	• 2 long-throw mono or stereo faders
• 1 short throw mono fader	• 2 EQ sections
• 1 EQ section	• 4 mono/2 stereo aux sends
• 4 to 6 mono aux sends	• 8 aux buses
• 4 to 8 aux buses	• 1 mono/stereo direct unput
• 1 mono direct output	• 3 effects loops, using line inputs
• 1 effects loop	• group/submix trim.
• group output trim.	

to the Crescendo.

Microphone/Line Inputs. Each I/O channel has two actively balanced mike preamps, with separate trim controls. They offer from -4dB to +60dB worth of gain; a pad is introduced automatically at low gain levels to minimize the chance of overload. Phantom power (+48VDC) is selectable, as are phase inversion and 100Hz highpass filter. You program these by pressing a preamp's BLOCK ATTENTION KEY, then using the command section.

Microphone preamp 2 also can be programmed to act as a group trim control (for one of the 24 group outputs); mike preamp 1 can be switched to trim the "submix" (a combination of the line inputs plus mike input 1). Mike preamp 2 acting as a group trim is handy: If in turn you select MIKE 2 as a source for one fader, that fader will behave like a group master fader. This is an easy way to use a group output as an extra aux effects send during mixdown.

The four +4dBm line inputs bypass the trim controls. Tape returns can be patched to any line input, but are normalled at the patchbay to LINE INPUT 4. In a manner that replicates switching between REPRO and INPUT on a tape machine, LINE INPUT 4 can be switched between TAPE and BUS (GROUP OUTPUT)—handy for comparing what's going to and coming off the tape deck.

Auxiliary Sends. There are two aux send "blocks" per I/O channel. Each block can receive as its source any or all of the two mike and four line inputs. Each block can feed any or all of eight aux output buses. Their levels may be pre- or post-fader, upper or lower level. Additionally, each of the two blocks can operate in one of 11 mono or stereo modes, including MONO, MONO PAN CENTER, MONO (VARIABLE) PAN, STEREO, and STEREO REVERSE. These same modes are available for faders, and are selected via the data entry wheel.

In other words, the Crescendo offers a wide variety of aux send options, for creating musicians' cue sends, effects sends, or a stereo mix that's separate from what's happening at the faders. This is great for broadcast and film/video post applications, where multiple mixes are often necessary.

Unfortunately, you're limited to a maximum of four mono or two stereo independent mixes per I/O channel (even though you can feed up to 8 aux buses at once)—eight mono or four stereo sends would have been ideal. With today's preponderance of effects, this is a minor shortcoming in mixdown and post-production work. It's a serious shortcoming if you're working with a large group of musicians who each demand individual mixes of wet and dry signals. In this case, one of the new headphone systems from Nemesis and others (which let musicians create their own mixes from channel direct sends) may be the way to go.

Channel Faders. The upper and lower fader sections are similar, in that both can receive any or all of the two mike and four line sources. Imagine controlling a 4-output source, like the Korg M1, with just one fader!

The upper and lower faders work in one of 11 mono or stereo modes, and *both* faders are 100mm long—a big improvement on the 60mm tape-monitor faders most boards have. Each fader section also has its own PAN/BALANCE knob, a CHANNEL ON/OFF switch, and both PFL (pre-fader listen) and AFL (after-fader listen) stereo solo switches. There also is a BLOCK ATTENTION KEY for each fader section to program sources, modes, and assignments.

You can assign either fader section to any or all of the two stereo outputs and a direct output (mono or stereo). The difference between the two sections is that only the upper fader has assignments to the 24 group outputs. (The traditional I/O board has group-out assignments for the lower fader but not the upper.) During a session, the upper faders could route synths, mikes, and other sources to tape, and rarely would need adjusting. While tracks are being cut, a monitor mix of previously recorded tracks could be made with the lower fader. In most cases this mix would be very similar to the final desired mix, which most engineers will prefer to do using the lower (closer) faders whenever possible. Again, another smart touch.

The Equalizers. The ATTENTION keys for the two equalizers are located within the lower fader section. Each EQ has four bands: low-frequency shelving, low-mid parametric, high-mid parametric, and high-frequency shelving. The two 4-band equalizers can be linked for stereo operation with one of the two faders, or can be used separately with each fader.

The equalizers are great. Frequencies, levels, and bandwidths appear in the alphanumeric display. You adjust settings by choosing a band (and its associated parameters) with an EQ BLOCK ATTENTION KEY, then vary the settings by spinning the data entry wheel. I love the fact that frequencies can be read in Hertz, or as musical note values (i.e.: *F#4*)!

Best of all is the MixView display: The computer displays EQ curves in a far more descriptive fashion than is possible on the alphanumeric display built into the mix controller. With different colors representing different EQ bands, and 1/4dB visual resolution, you'll be hard-pressed to return to using just *knobs* to set an equalizer.

The EQ sections sounded extremely clean, without any audible "warmth" or "character" beyond altering the frequencies at hand. Their sound may not delight everyone, but I suppose that's what $6,000 outboard EQs are for. Still, I found their sound neutral and predictable.

The most remarkable trait of these EQs is their consistency. I copied one channel's EQ settings to several other channels, fed them all the same program, and soloed at random. It was impossible to hear *any* difference between the different channels of EQ. Try that on a conventional board, with manually set controls!

Unfortunately, in some circumstances I could hear the dreaded "zipper noise" inherent in current DCA designs. At *extremely* high monitor levels, with little or no signal present, it is possible to hear faint 1/2dB "ticks" as an EQ band is cut or boosted—particularly at narrow bandwidths with a lot of boost. What does this mean in real-world studio terms? If the equalizer is set and isn't being adjusted, you won't hear zippers. Likewise, there is no zipper effect as EQ settings change between preset snapshots. With broadband, properly recorded signals, I doubt anyone would hear zippers as EQ settings are changed. The problem applies mostly to dynamically automated EQ changes, but overall, it's not a major concern.

Meters & Monitoring. The Crescendo's metering is the most comprehensive I've seen. Each I/O channel has two 19-segment, 3-color LED meters. Each can display any of the input sources, its associated BUS (group) output, or the levels of knobs and both faders (for automation). Six clip points per channel may be programmed, and meters may be switched between VU and peak (console-wide only). One mode—which lets one meter show group outputs and the other show tape returns—can be used as a quick check for tape recorder calibration before a session, particularly if you feed the deck various tones from the Crescendo's internal oscillator.

All monitoring functions are controlled from the master module. As described earlier, the three programmable monitor sections can select from various sources. Also, a headphone output under the controller can follow any of the three monitors.

> **Data sent between the Crescendo controller and mainframe travels a two-way street. Computer freaks will be impressed to know that the system's parallel interface operates at a 2MBaud rate, and that the controller itself is theoretically capable of operating at 8MIPS (*million instructions per second*)—and that's using ten-year-old Z80 chips!**

The talkback section is programmable to any of 12 destinations, and sports a built-in condenser mike. The oscillator is part of the Crescendo's talkback system, and is fully sweepable from about 20Hz to over 20kHz. However, as the oscillator sweeps upward through its three ranges, there are three mild thumps and temporary level drops. Like the EQ frequency setting, the oscillator can read its frequencies in Hertz or musical note numbers. What a bonus: The Crescendo doubles as a tuning fork!

The Crescendo also offers two primary stereo solo modes: a nondestructive mode that lets you solo channels and aux sends at will, without interrupting a mix. An "in-place" mode is "destructive"; it will interrupt a mix. Channel faders offer both PFL (*pre-fader listen*) and AFL (*after-fader listen*). Unfortunately, the eight aux master sends have no dedicated AFL solo buttons; you must configure one of the monitor sections to listen to an aux send.

The master module is also home to the board's two long-throw stereo faders. Most people will mix to just the ST1 fader; the second is usually used for Dolby Surround mixing or an alternate stereo send.

PROGRAMMING & AUTOMATION

All settings in the Crescendo can be stored with a few keystrokes as a *snapshot*, and can be recalled with a keystroke in 1/30 of a second. While snapshots can be programmed to trigger from external time code, they are not the ideal means of performing real-time automation of levels, EQ, and the like. Crescendo's *dynamic automation* offers the best way to automate such variable functions.

Snapshots. Too bad SSL owns the phrase "total recall" as a trademark, because this is the real thing. Instead of offering pictures of variable-control settings (which still require resetting gain levels, panning, and so forth, by hand), the Crescendo can store and recall *every* level, *every* setting, *every* routing. This is invaluable, since you can archive a session onto floppy disk (or the board's internal memory), then recall a complete mixing or tracking session with the touch of a button. Want some details?

Every channel of the board can store up to 99 snapshots of all its settings (including its two faders, EQ, etc.). While these snapshots can be stored and recalled on a per-channel basis, most of the time it's preferrable to store and recall them on a console-wide basis.

The snapshots worked flawlessly in "manual" mode—recalling them at will from the command module. I tried locking snapshots to time code (all formats), and everything worked perfectly. It's possible to program snapshot changes "on the fly" to time code, or they can be programmed for recall at cue points before the tape rolls.

The Crescendo can read external SMPTE/EBU time code in all formats, by the way, or can generate it itself to program and audition snapshot changes without tying up an external machine.

All faders and variable knobs have associated NULL lights. When you recall any snapshot, you can find a knob's preset level by moving it until the NULL light changes from red to green. (Faders have three NULL lights—"up," "down," and "aligned"—to show which direction the fader must be moved to match its position to the corresponding programmed level.)

Full Dynamic Automation. I

tried my hand at dynamic (real-time) automation, although at the time of this writing, Euphonix was fine-tuning the automation. So we'll touch on the working basics.

If you're familiar with traditional VCA-type fader automation, you'll recognize much of the Crescendo's dynamic automation features. Referenced to time code (external or internal), you can move faders as desired; when the code plays back, you hear the moves as you made them. You can "update" any fader—to alter a mix already in place—by moving it. (The current software is set so the update begins once you cross the NULL point.) One useful, fun touch is the alphanumeric display's "tape reel," which spins when incoming time code is rolling. At presstime, modes for "linking" faders (to raise the overall gain of a series of prerecorded moves) and cutting-and-pasting segments of moves were not yet implemented.

Automation of EQ settings is really impressive. It's a blast watching the computer monitor vary the EQ curves, locked to time code. Keep in mind, however, that zipper noise remains a possibility under automation, as it is during manual control.

Current software lets you program up to ten automation location points, to which you can "fast-forward" or "rewind" during a mix. In fact, each point is one of the console-wide snapshots. This is handy for those weird, trying mixes that require drastic changes, such as a radio drama that moves rapidly from scene to scene.

I look forward to seeing Crescendo's automation interface in a more finalized form.

FINAL SAY

The Crescendo is a powerhouse console. As you can judge from the number of exclamation points in this review, I'm blown away by its capabilities. Also, its future software potential is impressive: the faders could be configured as MIDI controllers, and EQ curves could be "painted" via mouse. Euphonix intends to include track sheets in the MixView software, and offer system updates via floppy disk or modem. But, as with any product, buy based on what it does today.

The Crescendo is not for everyone. As mentioned, the current software involves inconsistent, potentially confusing data-entry methods. The surface graphics need refinement. Some people will turn up their noses at possible DCA-induced zipper noise during automated EQ sweeps (though most users will never notice it). Others will want moving faders, and 32 or 48 groups instead of 24. And some people just won't feel comfortable working at this board: Whereas an experienced engineer can learn a new "conventional" console in 15 minutes, he or she must budget two to six hours for harnessing the Crescendo. Some studios and engineers can't afford that sort of "ramp-up" time.

For people who require total automation—and who are visionary enough to realize the Crescendo's full potential—this console is as close to perfection as one gets for less than $400,000. It sounds clean and quiet, and requires minimal control-room real estate. But most of all, the Crescendo's forté is incredible mixing power. I have seen the future of mixing, and it looks a lot like a Crescendo.

—*Brent Hurtig*

Thanks to Poolside Recording Studio of San Francisco for studio time with their Crescendo mixing console.

Shure VP88 Stereo Microphone

FOLLOWING THE popular trend in mid-side stereo condenser microphones ("MS" configurations), Shure Brothers has created the VP88, a contender that is compact, light and affordable. Originally marketed for the video production industry, the VP88 recently has crossed over into music recording and audio post-production.

Overview. This all-in-one microphone records audio information in MS or XY format. The former passes the incoming audio straight through in MS; an MS decoding box (not supplied) is required on the back end. The XY format encodes the audio in stereo, using one of three spatial settings: LOW STEREO (approximately 90°), MEDIUM STEREO (approximately 120°), and HIGH STEREO (approximately 150°).

In situations where vibrations and rumble present a problem, a low-frequency switch rolls off the signal below 80Hz at 12dB per octave. The low-frequency switch, stereo controls, and power supply are located in the body of the microphone, which alleviates the hassles of carting around separate boxes for MS encoding and power supply. (The microphone also can accept an external phantom supply between 9 and 52VDC.)

A nice surprise in the VP88 design is its cardioid pattern for the mid element rather than the typical hypercardioid pattern. A phone call to Shure revealed the reasoning behind this: The cardioid pattern has more rear-signal rejection than the narrower hyper-pattern. This is extremely useful for cancelling out unwanted signals from behind. You can purchase an optional shock-mount package to attach the VP88 to a camera or a fish- or boom-pole.

Finally, and conveniently, Shure supplies a single 5-pin cable that splits to two XLR pigtails at your recorder rather than at the microphone.

Listening Tests. I was painting a wall in my house when it began to rain, the thunder cracks echoing back and forth through the canyon. This presented an opportunity to try the VP88 on a sound effect containing full frequency range with lots of dynamics. I ran outside with my portable Sony TCD-10 Pro DAT recorder and the VP88 to record 30 minutes of rainstorm. The next day in the studio, I was delighted with the recording's stereo reproduction, extreme realism, and fantastic stereo imaging. I also was impressed by the complete lack of microphone noise.

In another listening test, I recorded an acoustic guitar with the VP88 placed midway between the soundhole and the neck/body joint. This yielded a wonderful stereo guitar image. By changing the microphone's angle and the stereo spatial imaging, I was able to coax different stereo widths and tonal timbres from the instrument. (One positive feature of MS recording is that when you mono an MS stereo track, the sides cancel out, thus leaving only the

Product: Shure VP88 Stereo Microphone
Manufacturer: Shure Brothers Incorporated, 222 Hartrey Ave., Evanston, IL 60202; (708) 866-2200.
Price: $995.
Size: 11-5/16" x 2-1/2".
Weight: 15.5 oz.
Frequency Response: 40Hz - 20kHz.
Dynamic Range: 105dB (A-weighted).
Max. SPL: 120dB into 800Ω.

mid-cardioid signal.)

We also conducted two separate distant miking tests—one on a baby grand piano and one on a guitar amp. With the VP88 I could capture the sound of the instruments in the room, providing a "you are there" impression for the listener. In both instances I also attained pleasing results by using a (mono) dynamic microphone up close combined with the spatial imaging of the Shure VP88 placed in the room.

Accessories. The VP88's $995 price includes battery, carrying bag, foam windscreen, swivel adapter, and a Y splitting cable. Options include a shock-mount rig ($175) and the 5-pin 25' extension cable ($75). Shure also sells a Rycote windshield accessory line for the VP88, including a shotgun pistol grip, sound zeppelin for light wind noise, wool cover for moderate wind noise, and a "Wooly" for heavy wind environments.

Conclusions. The VP88 is a portable, quiet, good-sounding microphone for those who want single-point stereo at a moderate price. (Many of its single-point stereo competitors list at over $2,000, not including power supply.) If you're recording in the field, I recommend investing in the 5-pin extension cable so you won't have to deal with two cables extending from the microphone. The optional shock-mount package also is a must: Shure has created one of the best shock-mounting systems around; it can interface to almost all mounting formats.

Compared to Crown's SASS-P microphone (see *EQ* Mar/Apr '90), the VP88 presents a more up-front, "in your face" sound. For pure ambient or background recording—or perhaps for concert hall recording—the SASS-P may be a better choice. Furthermore, with a relatively low 150° maximum spread, the VP88 may not be able to create an "unnaturally" wide stereo image (as can some other stereo microphones). But for clean, all-purpose miking that sounds highly natural, the VP88 is impressive.

Since no two people hear the same way, a judgment of microphones is highly subjective. Nonetheless, if you're in the market for a stereo microphone, a test drive of the VP88 is most definitely in order.

—Scott Martin Gershin

The Art Of Squash

EQ Examines Three New Comp/Limiters: Aphex Expressor 651, Orban 412A, & JBL/UREI 7110.

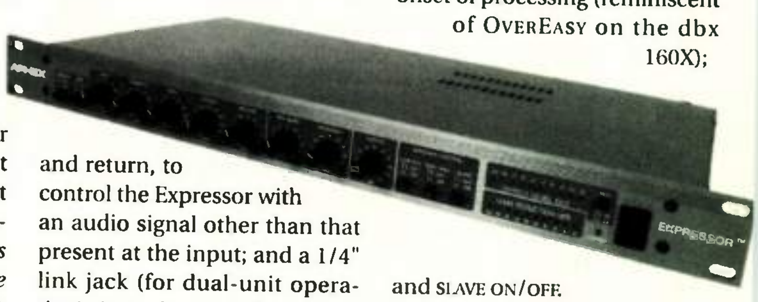
WHILE SOME ENGINEERS abhor compressors and limiters, in search of the elusive "transparent" sound, most of us use them for an intentional effect to reduce the dynamic range of an instrument or vocal. Sometimes this effect is subtle: to sustain a bass by apparently boosting low-level signals as the strings decay, for instance, or to act as a safety "limiter" to prevent occasional overloads from a vocalist.

Other times a flagrant amount of comp/limiting may be in order. When we mixed Dave Edmunds' new album *Closer To The Flame* here at Capitol Records, for instance, he liked his vocal sound heavily limited, and we used an older UREI 1176LN to do it. What went into that limiter was not what came out; that was the intent. [*Ed. Note: For theory basics on compression and limiting, see* "EQ *Background Notes," page 73, Mar/Apr '90.*]

But the 1176LN has become somewhat of a collector's item, as have the classic Teletronix LA2A and UREI LA3A, among others. If you're looking for a modern "squash box," three new comp/limiters—the Aphex Expressor 651, Orban 412A, and JBL/UREI 7110—are designed to deliver. They are not "do everything" dual-channel compressor/limiter/gate/expander dynamic processors (such as the Rane DC24; see the Mar/Apr '90 EQ). Rather, these three single-channel, single-rackspace units are dedicated to compression and limiting. *EQ* evaluated them by using them as conventional compressors, and also to squash and limit the daylights out of stuff—just to see how what went in came out!

APHEX EXPRESSOR 651

The Expressor's rear-panel connections include XLR inputs and outputs; 1/4" sidechain access for send and return, to control the Expressor with an audio signal other than that present at the input; and a 1/4" link jack (for dual-unit operation). According to Aphex, either pin 2 or 3 of the XLR connectors may be wired hot, due to the company's approach to active balancing (ins and outs must be consistent to avoid phase inversion).



Front-panel controls include: PROCESS ON/OFF for bypass that disengages the output control; INPUT (±15dB); THRESHOLD (-20dB to +20dB); RATIO (1.1:1 to 50:1); ATTACK (0.05 to 100ms); RELEASE (4ms to 4s); and OUTPUT (-12dB to +18dB). Two LED displays are provided: The OUTPUT METER shows in VU increments, and the GAIN REDUCTION METER shows in 2dB increments (both operate regardless of the PROCESS ON/OFF setting). I liked these meters because they really showed how my adjustment of the controls affected the signal. However, I did not find them accurate when I sent in a 0VU tone; this was confirmed in our Lab Test.

Front-panel buttons include LOW CUT ON/OFF, which inserts a low-cut filter at 80Hz into the sidechain; SOFT KNEE ON/OFF, in which gain reduction becomes more gradual and less apparent than the standard "hard knee" onset of processing (reminiscent of OVEREASY on the dbx 160X); and SLAVE ON/OFF.

Here things get a bit confusing. With SLAVE ON, the Expressor is controlled by another (master) Expressor's settings, regardless of the gain reduction setting. With SLAVE OFF, two Expressors can be linked, and the one exhibiting the most gain reduction will control the other.

EQ Lab Test

Compressor/Limiters
Product: Expressor (Model 651)
Manufacturer: Aphex Systems Ltd., 13340 Saticoy St., North Hollywood, CA 91605; (818) 765-2212.
Price: $495
Operating Level: +4dBm.
Maximum Output Level (@ +4dBm input):
+30.3dBm (26dB gain); with input and output level controls set to "0" and a +4dBm input, meter indicates -3VU.
Frequency Response (@+4dBm, unity gain):
Claimed: 5Hz to 100kHz (+0, -0.2dB).
Tested: 20Hz to 40kHz (±0.3dB).
THD (@+4dBm, 1kHz, unity gain):
Claimed: 0.006%.
Tested: 0.004%.
Signal-To-Noise (@+4dBm, unity gain):
Claimed: -85dB.
Tested: -92.3dB; -75.0 below operating level @ max output.

So if two Expressors are linked by a cable, you'll need to disconnect the cable when you want independent operation; you may want to bring the link connections up on your patchbay. We would have preferred a front-panel LINK switch.

Two Enhancement Circuits. The SPR button (introduced in the Aphex Aural Exciter Type III) controls a "Spectral Phase Refractor" circuit that is independent of other controls. According to the owner's manual, the phase of the low-frequency audio spectrum becomes delayed compared to the mid and high frequencies, during the normal steps involved in recording, mixing, and mastering. With SPR switched on, the bass frequencies (up to 150Hz) are processed to correct this delay and "restore clarity and openness." Perhaps this is so, though neither myself nor several colleagues in the studio at Capitol Records were able to hear any difference between a "straight" electric bass and an "SPRed" bass. (The SPR function is intended to be most noticeable with broad spectrum signals.)

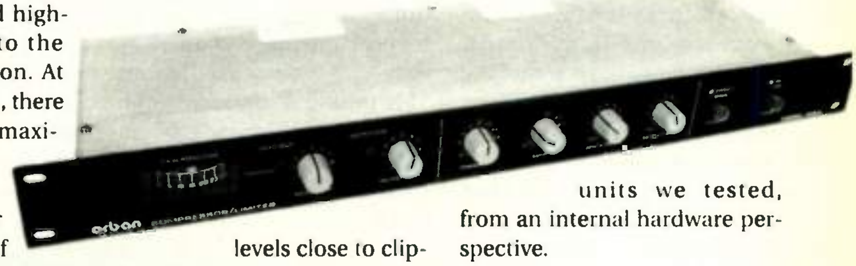
The Expressor also includes a high-frequency ratio control (HFX), which automatically adds EQ, but only when necessary. Since perceived high-frequency "brightness" is diminished the more a signal is compressed, this circuit lets you add high-end EQ proportional to the amount of gain reduction. At the minimum ratio of 0:1, there is no HFX effect. At the maximum setting of 1.0:1, there is 1dB of high-frequency expansion (or boost) for every 1dB of gain reduction. A frequency control lets you adjust the high-pass corner shelving frequency from 2kHz to 20kHz. Those engineers accustomed to running their comp/limiter's output to an EQ will be pleased with this internal automatic compensation control. The HFX circuit is a smart addition—especially for those in live sound, where full-time high-end EQ boost generally means more feedback.

The owner's manual generally is well-written, and provides several sample setups for various instruments.

ORBAN 412A

The Orban 412A has input/output connections made via barrier strip (with screw-down terminals) or 1/4" phone jacks. (Out test unit had an optional XLR connector kit; as of July 1990 XLRs are being shipped as standard equipment.) Sidechain in/out connections are made via 1/4" phone jacks. The input is floating, active-balanced, as is the output.

Front-panel controls include: INPUT ATTENUATOR, which adjusts the level of signal available to the compressor and determines the amount of gain reduction, depending on the threshold setting; THRESHOLD (±10dB) with a "0" detent of 25dB of available gain reduction; RATIO (2:1 to ∞:1); ATTACK (500us to 200ms); RELEASE (3dB/sec to 80dB/sec); SYSTEM BYPASS switch; and a gain-reduction meter. The gain-reduction overload lamp indicates maximum gain reduction at levels close to clipping. An OUTPUT ATTENUATOR includes a lamp which indicates output clipping.



Unconventional Operation. The owner's manual suggests setting the 412A's controls differently from what most engineers are accustomed to. For a relatively constant output level, Orban suggests you keep the THRESHOLD control at the "0" detent position and use the input attenuator to achieve the desired gain reduction. Also, the ATTACK control affects the threshold level as well; as you make the attack time slower, you reduce the threshold of compression. This is most unusual.

I highly recommend that people read the manual before trying to operate this compressor—but I don't think this is a big selling point in its favor. I don't know many engineers who have the time or the inclination to read a manual before being able to operate the 412A as a straight-forward comp/limiter. To its credit, the 412A is perhaps the most well-built of the three units we tested, from an internal hardware perspective.

JBL/UREI 7110

The 7110 has a recessed front panel to prevent the controls from being changed accidentally. We like the fact that the inputs/outputs consist of XLR connectors (pin 3/hot), 1/4" phone jacks, *and* barrier-strip terminals. Input can be balanced or unbalanced; output is unbalanced, but can be converted to floating balanced with an optional output transformer.

The 7110's rear panel provides an additional 1/4" phone

EQ Lab Test

Product: 412A
Manufacturer: Orban c/o AKG, 1525 Alvarado St., San Leandro, CA 94577; 9415) 351-3500.
Price: $459
Operating Level: +4dBm.
Maximum Output Level (+4dBm input):
+18.5dBm (14.5dBm gain); overload lamp illuminates @ +14.5dBm.
Frequency Response (@+4dBm, unity gain):
Claimed: 20Hz - 20kHz (±0.25dB).
Tested: 20Hz - 40kHz (±0.2dB).
THD (@+4dBm, 1kHz, unity gain):
Claimed: <0.05%.
Tested: 0.006% (output set to maximum).
Signal-To-Noise (@+4dBm, unity gain):
Claimed: >85dB below operating level.
Tested: -84.6dB; -67.9dB below operating level @ max output.

IN REVIEW

EQ Lab Test

Product: 7110
Manufacturer: JBL/UREI Professional, 8500 Balboa Blvd., Northridge, CA 91329; (818) 893-4351.
Price: $495
Operating level: +4dBm.
Maximum Output Level (+4dBm input): +25dBm (21dB gain); with a +0.0dBm input, meter indicates 0VU.
Frequency Response (@+4dBm, unity gain): *Claimed:* 20Hz to 20kHz (±1.0dB).
Tested: 20Hz to 40kHz (±0.1dB).
THD (@+4dBm, 1kHz, unity gain): *Claimed:* N/A.
Tested: 0.005% at operating level, 1kHz.
Signal-To-Noise (@+4dBm, unity gain): *Claimed:* N/A.
Tested: -93.0dB; -74.8dB below operating level @ max output.

jack labeled deceivingly as the DETECTOR INPUT. I say that since it really has nothing to do with the DETECTOR circuit (described below). It is, in fact, a "key" or "sidechain" input, to drive the compressor from another source. It should be labeled as such, to avoid confusion.

The 7110's front-panel controls are well laid-out. These include: THRESHOLD (-40dB to +10dB); ATTACK (1ms to 50ms); RELEASE (50ms to 2s); RATIO (1.5:1 to ∞:1); OUTPUT (±20dB); switch-selectable, green LED display for output or input level; red LED display for gain reduction; DETECTOR control (more on this later); AUTO switch to preset the DETECTOR, ATTACK, and RATIO controls and engage a program-dependent RELEASE circuit; and LINK to strap multiple units together. The LED displays are extremely accurate.

Two In One? The DETECTOR circuit allows the 7110 to respond as an average compressor *and* a peak compressor. Using it is like having a second threshold control. With the control set to AVERAGE, the threshold for peak-limiting is 20dB higher than that for average limiting. This setting works well for vocals, some synths, and other instruments that don't have many transient characteristics, and it gives you some protection against that unexpected peak. As you rotate the control toward the PEAK setting, the threshold for peak signals is reduced progressively, giving you more control over the actual peak-to-average ratio of your program material.

The AUTO mode provides a simple way to use the limiter when you want some protection and don't have the time to set individual controls. When AUTO is selected, an LED lights up (something I always wanted the dbx 160X to do for its OVEREASY button).

COMPARISON RESULTS

For a subjective and unscientific listening test, I set up our three devices along with three classic "benchmarks": the Teletronix LA2A and the UREI 1176LN and LA3A. Program material included kick, snare, a Marshall-sounding electric guitar solo, electric bass, and a lead vocal. Finally, joined by several colleagues, I listened.

The best-sounding of the three new compressors? The JBL/UREI 7110. Its overall sound, particularly with kick and snare, was better than the other limiters. In fact, the kick sounded better processed by the 7110 than it did with no compression. The detector circuit worked well.

The Aphex Expressor came in a close second in subjective listening. As a "squashing" device, it performed extremely well. I used the Expressor on an actual mix to limit a vocal heavily, and employed the output as an effects send for echo; it worked great. While we were unable to hear the effect of the SPR, the HFX circuit is a unique touch. For many users, these additional features may make the Expressor a more appealing purchase than the 7110.

The Orban 412A came in a distant third, basically because we didn't think it sounded good. It was noisier and degraded the transient response of the program material too much. Furthermore, its unusual operational design makes it that much less appealing.

There are reasons professionals still use the older "classic" comp/limiters—they sound great with lots of different program material, are easy to use, and work for light compression or heavy limiting. The JBL/UREI 7110 will fit the same bill for many people; others will prefer the Aphex Expressor. It may not be necessary to re-invent the wheel, but new features certainly make for a smoother ride.

—Leslie Ann Jones

Fostex 2016 Line Mixer

WITH THE INCREASING availability and use of multi-timbral/multi-output MIDI sound modules—many with built-in EQ and digital effects—it's easy to find yourself in a situation where you have more outputs from sound sources than your mixer has inputs. Instead of buying a new mixer with more inputs, you can solve the problem with a *line mixer*. Line mixers provide a way to submix numerous inputs down to a stereo pair that can be routed to channels or aux inputs on a main mixer. The Fostex 2016 is a recent entry into this line-mixer marketplace.

Overview. The 2016 is a 2-space rackmount unit designed as two separate 8x2 mixers, each with two post-fader aux sends and associated returns. However, the designers included a few not-so-standard features. For instance, you can chain the two 8x2 mixers into a 16x2 mixer, with or without chaining the aux sends, by pressing the LINE LINK button on the front panel. The aux sends optionally can be linked by pressing LINK AUX. In this mode, aux sends 1 and 3 and aux sends 2 and 4 are chained.

In addition to the 16 line inputs and four returns, the 2016 offers four line bus inputs and four aux bus inputs. We count more than 16 inputs here. Would you believe 28? That's right: 16 with panning, aux sends, and gain control, and four more with level adjustments. The remaining eight bus inputs have no associated controls, yet are ideal candidates for creative patching.

The unit's rear panel contains connections for four line

bus inputs and four aux bus inputs (RCA jacks); four line outs, four aux sends, and four aux returns (1/4" phone jacks).

The front panel provides a pair of 5-segment LEDs for each 8x2 section. The owner's manual describes these LEDs as level meters, but it's not clear as to what units the numbers printed next to each segment are counting—dB? VU? dBm? We had to phone Fostex for the answer: VU, -10, -5, 0, +3, and +6.

The layout of the front panel is clean and simple, although the 72 knobs make it a bit cramped for large hands. Each of the 16 inputs has two pots labeled AUX SEND LEVEL, a BALANCE pot, and a GAIN pot. Additionally, there are four AUX SEND pots, two STEREO RETURN level pots, a MASTER LEVEL pot, and a LINE LEVEL pot.

The front panel also provides a 1/4" phone jack directly below each GAIN control. Front-panel inputs take priority over the rear inputs; however, using them is a bit awkward when trying to make precision, on-the-fly adjustments. The front-panel control pots are small to begin with, and their overall feel is toy-like. Even with nothing plugged into the front panel, making adjustments can be difficult. This type of arrangement lends itself to setting levels at the beginning of a session, then leaving them alone.

The Mixer At Work. Remember all those MIDI sound modules in need of a place to plug in? Well, here it is, complete with aux sends. Now you can connect all those sound sources through the line mixer and get two (or four) effect sends. Once the levels are set, you can control the volume of the individual timbres on the sound module by using MIDI controller #7.

When used in this configuration, the 2016 works and sounds great, adding no noticeable noise. In fact, its overall sonic performance is great, as the *EQ* Lab Test concluded. It is key to take the time to adjust the various volume levels and gain levels until you find the optimal setting for each unit. One limitation is that this mixer is designed with -10dBv nominal levels in mind. You can use it in a +4dBm setting, certainly, but the 2016's limited headroom and maximum output level would require paying close attention to signal levels.

The owner's manual illustrates several different patch configurations, including one that shows how to use your mixer's direct outs and the 2016 to gain two additional effect sends. The one we liked best involves using the 2016 as two separate 8x2 mixers with two aux sends apiece (not much different from the configuration described above; the main difference lies in *not* chaining the aux sends). This way we could use one stereo effects setting for inputs 1 through 8, and a different stereo effects setting on inputs 9 through 16.

We're impressed with the flexibility afforded by the 2016. We can't overlook the fact that it offers 16+ inputs at a price competitive with 8-input line mixers. Just think of all the possibilities presented by using *two* 2016s.

Conclusion. The Fostex 2016 is an affordable, good-sounding, flexible line mixer, offering up to four aux sends, although only two are available to each 8x2 section. We would be willing to trade the 1/4" phone jacks on the front panel for larger gain knobs; still, the fact that the inputs are equipped with phone *and* RCA jacks is a nice touch. The compact layout, clean sound, and abundance of inputs makes the 2016 a good choice for MIDI mixing applications, and for adding cue sends and/or effect sends to an existing setup.

—Tom Ford

EQ Lab Test

Product: 2016 Line Mixer
Manufacturer: Fostex Corp. of America, 15431 Blackburn Ave., Norwalk, CA 90650; (213) 921-1112.
Price: $399.
Frequency Response:
Claimed: 20Hz - 20kHz.
Tested: 20Hz - 40kHz (±1.7dB).
Signal-To-Noise:
Claimed: 80dB.
Tested: 107.5dB.
Total Harmonic Distortion (@ -10dBv input):
Claimed: 0.01% (100Hz - 10kHz).
Tested: 0.01%; 1.0% @+11dBv input.
Crosstalk (@ 1kHz):
Claimed: 65dB.
Tested: >65dB.
Maximum Output Level:
Claimed: n/a.
Tested: -3dBv.

ADVERTISERS INDEX

AES....64
Akai/IMC....59
Amek....Inside Front Cover
Ampex....13
Aphex....50
Audio Institute....33
Audio-Technica....Inside Back Cover
BBE....66
Brainstorm....33
Bruel & Kjaer....31
Crown....44
D&R....49
DOD....5
Drawmer....26
EQ Subscription....62
First Light....67
JBL....Back Cover
Jeanius Electronics....33
Lexicon....First Inside Page
Lone Wolf....55
Opcode....6/7
Peavey....27
QSC....2
Rane....46
Recording Workshop....58
Roland....18/19
Shure....11
Soundtracs....21
Sunkyong....43
Tascam....40/41
The DAT Store....58
Yamaha....28/29

REFLECTIONS

◄◄ PAGE 8

Qs, but these colorations detracted from the original material. Our test marketing with earlier models revealed that engineers want monitors that tell exactly what the mix is doing and take the guesswork out of equalization.

The reference studio monitor should be the ideal transducer, pass on exactly what is put in, be as sonically transparent as possible. You should be able to hear your mike placements, possible polarity inversions, dynamic overloads, etc., while providing a true point source for stereo effects. It should be free of phase error, possess very low if any overall group delay between drivers, have sharp energy vs. time curve, be not only flat and smooth in response, but fast in transient response, have low cabinet diffraction, and provide accurate representation of consumer-applicable sound standards.

Mixes done on colored monitors stand less chance of holding up on a wide range of playback than a more neutral monitor. There is also the increased possibility of exaggerated acoustic effects. With a neutral monitor, the engineer can tailor work to a specific market, knowing exactly how much color has been added. Many studios today run room and speaker curves to establish a reference point. Subjective listening alone does not seem to be a very critical or demanding acid test.

The control should have been your own source material, so the various changes each monitor added in playback could be identified clearly, instead of comparison playback with highly colored monitors. A better test may have included a comparison of what kind of response the listeners liked and the actual frequency response curves. If the response curve and/or energy time curve of the reference is not known, then how can the listeners know where they're starting from?

A prime example of the confusing results is the flat response averages. Our LS261 with the midboost switch on was judged flatter than when in the accurate position. We can assure you this is definitely not the case. In addition, many perceived imaging effects can be linked to efficiency differences as the apparent dynamic range (the difference between lowest lows and highest highs) is exaggerated when there is a lack of level compensation.

With the recording world becoming ever more complex, there is a need for more demanding reviews of products. We would like to suggest a true acid test covering all facets of this complicated issue of near-field monitoring.

JASSA LANGFORD, DESIGNER
Digital Designs
El Centro, CA

Mr. Langford raises a number of excellent points. But as we stated in the original article, one reason many engineers use NFMs—rather than equalized, flat-as-possible, open-field monitors—is that most "offer a closer representation of the average speakers found in real-world listening environments." Many manufacturers claim conflictingly that their NFMs are accurate and *"representative." That's why we chose to skip the test gear and just listen. We used prerecorded material, since our own source material would have been colored anyway by the studio gear and monitors we used to record our tracks. And to clarify a question raised by others: Each monitor pair was listened to in an optimum near-field position, one pair at a time, in relation to the control pair. We did not set up a "wall of sound," sit in one place, and monitor speakers at random. (Regarding the Digital Designs LS261s, see For The Record, page 17).* •

UPDATE CONTACTS

AMS Industries, Billington Road, Burnley, Lancs. BB11 5ES, UK, (0282) 57011 [in U.S.: 1180 Holm Road., Suite C, Petaluma, CA 94954, (707) 762-4840]; **Alpha Audio,** 2049 W. Broad St., Richmond, VA 23220, (804) 358-3852; **Amek,** Regent Trading Estate, Oldfield Rd., Salford M5 4SX, UK, (061) 834-6747 [in U.S.: 10815 Burbank Blvd., N. Hollywood, CA 91601, (818) 508-9788]; **AES,** 60 E. 42nd St., New York, NY 10165, (212) 661-8528; **DDA,** Klark-Teknik, 30B Banfi Plaza North, Farmingdale, NY 11735, (516) 249-3660; **Digital Designs,** 125 W. Main St., El Centro, CA 92243, (619) 353-1290; **Dolby,** 100 Potrero Ave., San Francisco, CA 94103, (415) 558-0200; **Drawmer,** Charlotte St. Bus. Ctr., Wakefield, W. Yorkshire WF1 1UH, UK, (092)-437-8669 [in U.S.: Quest Marketing, 2245 Commonwealth, Auburndale, MA 02166, (617) 964-9466]; **Focusrite,** Unit 2, Bourne End Bus. Ctr., Bourne End, Bucks SL8 5AS UK, (0628) 819465 [in U.S.: 1100 Wheaton Oaks Ct., Wheaton, IL 60187, (312) 653-4544]; **GML,** 7821 Burnet Ave., Van Nuys, CA 91405, (818) 781-1022; **Kitsch Studios,** 15 rue Wéry, 1050 Brussels, Belgium, (02) 640-0880; **Lowery Assoc.,** Box 2021, Norcross, GA 30091, (404) 662-8504; **Motionworks,** Step Centre, Osney Mead, Oxford OX2 0ES, UK, (0865) 790-577 [U.S. Distributor: 21st Century Ltd., 2002 N. Beachwood Dr., L.A., CA 90068, (213) 463-4718; **Neve,** Berkshire Industrial Park, Bethel, CT 06801, (203) 744-6230; **New England Digital,** 49 N. Main St., White River Junction, VT 05001, (802) 295-5800; **Solid State Logic,** Begbroke, Oxford OX5 1RU, UK, (0865) 842300 [in U.S.: 320 W. 46th St., NYC, NY 10036, (212) 315-1111]; **Soundcraft,** JBL Professional, 8500 Balboa Blvd., Northridge, CA 91329, (818) 893-4351; **Studiomaster,** 950 Hampshire Rd., Ste. 102, Westlake Village, CA 91361, (805) 494-4545; **Tannoy,** 300 Gage Avenue, Unit #1, Kitchener, Ont. N2M 2C8, Canada, (519) 745-1158; **Thompson Audio Developments,** Box 373, Biggleswade, Bedfordshire SG18 0AQ, UK, (0767) 601655.

CLASSIFIED

How To Place An *EQ* Classified Ad

REGULAR CLASSIFIEDS

A **Regular Classified** is an ad with *text only* – no graphics or logos. You supply us with a typewritten copy of your ad (or a handwritten one – please print clearly). We typeset it for you.

RATES

$10.00 per line.
(Figure 30 lower case characters per line; 24 in ALL CAPS.)

OPTIONS:
Bold line: $10.00 extra.
Black border around ad: $10.00 extra.
Gray screen OR **knockout** (white print on black background): $10.00 extra.

Frequency Discount:
Prepay for four issues, get the fifth free.

DISPLAY CLASSIFIEDS

A **Display Classified** is an ad featuring your logo and/or artwork – *you* supply us with the camera-ready art. **All Display Classifieds must measure 1-5/8" wide.** Rate is determined by height:

RATES

$80.00 per inch in height
(always 1-5/8" wide)

Frequency Discount:
Prepay for four issues, get the fifth free.

DEADLINES

May/June issue: **February 23**
July/August issue: **April 23**
Sep/Oct issue: **June 23**
Nov/Dec issue: **August 23**
Jan/Feb issue: **October 23**
March/April issue: **December 23**

CATEGORIES

- Equipment for Sale
- Equipment for Rent
- Parts/Accessories
- Sounds & Software
- Tapes/CDs/LPs
- Instruction
- Employment Opportunities
- Business Opportunities
- Recording Services
- Repair/Maintenance
- Studios for Sale

THE FINE PRINT

All Classified advertisements must be prepaid by check, Money Order, MasterCard, or Visa.

Stated or implied discounts on new equipment are not allowed (Note: Manufacturers may revise their *own* prices.)

All Regular Classifieds must be submitted in writing via mail or FAX (please print legibly or type).

All Display Classified advertisers must supply us with camera-ready art by the deadline. Ads may be *re-run* via telephone, before the deadline, *only* if you wish to pay with a credit card, and are not making copy changes.

You may pre-pay for as many issues as you wish.

All ads must be related to recording/music/multimedia.

For our reference purposes only, your complete name, full address (PO Boxes not sufficient), and telephone number must be supplied with your order, whether or not it is included in the ad copy.

Ads received after the deadline will be held for the next issue unless otherwise specified.

NEED TO KNOW EVEN MORE?

Write:
***EQ* Classifieds**
P.O. Box 4032
Cupertino, CA 95015.

FAX: (408) 446-1088.

Or Call!
Ask for Amy Miller,
EQ Classifieds,
at (408) 446-1105
OR (800) 999-9018.

EQUIPMENT FOR SALE

DAT-ready coaxial digital output added to any CD player w/optical output – And most Technics w/o. $99.95. (801) 466-3196. Performance Audio.

All major recording gear and digital keyboards. Courteous service. You're only a free call away from total happiness. **MUSICMAKERS. (800) 395-1005.**

PARTS & ACCESSORIES

Portable Sound Panels

- Isolate specific areas
- Many sizes and options
- Panels start at $19.95/ea Complete w/foam

Write for free catalog

Island Cases
1121-I Lincoln Ave., Holbrook, N.Y. 11741
800-343-1433 • In N.Y. 516-563-0633

EMPLOYMENT OPPORTUNITIES

Planning An Audio Career?
Get New Ears!
A Guide to Education in Audio & the Recording Sciences. Over 80 schools profiled, 324 listed. The most complete reference available. 208 pages of info.
$11.95 post paid. $15.95 foreign air.
New Ears. 1033 Euclid, Syracuse, NY 13210
ph: 315-425-0048

Open Tracks: Editorial openings with the creative recording magazine. One position requires test lab experience. If you can express yourself on a Macintosh as well as a mixing board, send resume to: Editorial Director, *EQ*, 20085 Stevens Creek, Cupertino, CA 95014. EEO.

INSTRUCTION

BE A RECORDING ENGINEER
Learn Recording and Mixing, Signal Processing, MIDI, Digital Audio and more. Intern at a top NY studio and benefit from lifetime job placement assistance. Ask about the special summer program for college students and the new VIDEO TECHNOLOGY PROGRAM
800-544-2501
NY NJ CONN 212-777-8550
Lic. by NYS Education Dept / HS or GED Required
App. for Vet Training / **Financial Aid if Eligible**
Institute of Audio Research
64 University Place, Greenwich Village, New York, NY 10003

SOUNDS & SOFTWARE

Sampler Owners:
Professional quality digital samples on TDK SA-X chrome cassette! Hundreds of analog, FM and electronic synth sounds, effects, and stacks for your sampler. Each tape is a first-generation real-time transfer, direct from digital. Send check or M/O for $15.95 + $3.00 s/h to:
Evening Star Records,
PO Box 6264, Malibu, CA 90264.

EVENING STAR RECORDS
PROFESSIONAL AFFORDABLE SOUND SUPPORT FOR THE ELECTRONIC MUSIC INDUSTRY

OTHER

"EARRINGS FOR MEN." Special line designed for musicians. Solid sterling silver treble cleff or eighth notes earring (1), $8.95. Solid 14K gold, $59.95. Catalog $1.00. **DANGER FASHIONS,** 535 Grimsby, Henderson, NV 89014.

I Can't Believe My Ears

Why Some Professional Recorders Sound Worse Than Consumer Machines, And What We Can Do About It.

BY ROGER NICHOLS

AFTER TEN YEARS OF RECORDING digitally, I am finally getting used to the detail that is available in the digital domain. When I first started recording digitally, *anything* digital was so much better than anything analog that it didn't make a difference what brand of machine I used. I would rather record digitally with a Dixie cup and string than with an analog machine.

Now that I've made up my mind to record digitally forever, I can't decide which analog-to-digital converters to use. Every digital machine I listen to sounds different. Every digital machine manufacturer has a different philosophy about how to treat the conversion from the analog domain. Some concentrate too much on the bells and whistles added to the machine, and not enough on the quality of the audio it records. Some professional 2-track machines that cost over $30,000 do not record audio as well as some consumer DAT machines that cost under $2,000.

On a recent project I ended up using the converter section of a DAT machine for my analog signal, and patched the AES/EBU output into the professional machine. This sounded much better than the analog converter section of the pro machine. I might as well have mixed to the DAT machine and rolled the nice, cute, professional machine out into the street. The reason I didn't is because in Hollywood, there are street signs posted that say "TOW AWAY ZONE: NO DIGITAL MACHINES 9AM to 11AM MONDAY THROUGH FRIDAY."

Consumer DAT machines are right up there in the push for the ultimate technology in analog/digital/analog conversion. I've seen and heard consumer decks that boast resolutions of 18 bits, 19 bits, 20 bits, and best of all, one bit. Some incorporate oversampling inputs of up to 256 times with digital filtering. Output stages in other machines use two D/A converters per channel, to minimize zero-crossing distortion.

Professional digital machines are slow to catch up, however. The volume of machines produced for professional use is miniscule compared with the number of consumer DAT machines, CD players, and now, digital preamps and receivers hitting the consumer market.

Companies such as Apogee, Crystal, Digidesign, Wadia, Stax, Yamaha, and others are coming to the rescue of us professionals who are oversampling-poor and resolution-shy. With the exception of Digidesign and Yamaha (who also build their own digital recorders), these companies can keep their winning edge in the converter cold war by just supplying the devices necessary to perform the conversions, and letting someone else deal with getting the information on and off tape.

I don't want to mention any names, but Yamaha has a new 2-channel A/D converter box, model AD2X, with delta-sigma conversion, 19 bits of resolution with 64 times oversampling, AES/EBU and S/PDIF outputs, balanced analog input, and word sync in and out. I fed the box's AES output into the AES port of a Sony 3402 2-track machine. Then I mixed my 48-track digital master through a 72-input Neve VR-P console into the Yamaha box. The Sony 3402 can chase-lock to SMPTE time code without an external synchronizer, so I was able to lock the mix that I printed to the 2-track with the 48-track machine, so I could A/B the console mix with the playback of the 2-track. They were absolutely identical. Considering the box's price of $1,695, no studio should be without one. Now I know I never again will use the A/D converters built into the 3402.

Two years ago I got my hands on the prototype of a new A/D converter that Apogee will market soon. I had virtually the same experience. Its performance was head-and-shoulders above any of the converters built into the professional machines. This box should retail for around $1,000, but it's not out yet, so I'll keep you posted.

Wadia builds a D/A converter that sounds wonderful. The price is also wonderful: about $10,000 for two channels. Definitely worth it for a professional installation, but a bit steep for my home studio. Masterfonics in Nashville uses the Wadia box in their mastering rooms for monitoring all the digital-domain mastering.

This all boils down to the fact that we should make sure the machines we use in the studio sound at least as good as the CD and DAT players being used to play back the music we produce.

When you're trying to decide which digital machine to mix to, give a long, hard listen to the analog performance. If it is not quite what you think it should be, try an outboard converter box. This is not cheating; remember, all is fair in love and digital audio. •

Engineer extraordinaire Roger Nichols is master of the control rooms at Soundworks West in West Hollywood.

ILLUSTRATION: SCOT HALPIN

Grand sound... for less than 1/3 of a grand!

The impressive new AT4031 Studio Cardioid Capacitor Microphone.

Our new AT4031 makes a great first impression. And it builds from there. The sound is effortless and natural over the entire spectrum of voices and musical instruments. It's the result of wide, flat frequency response, low distortion, and excellent dynamic range.

Very fast transient response

The outstanding performance is the product of uncommon precision inside the fixed-charge element. The diaphragm is only 0.00016″ thick (just 4 *microns*), including the ultra-thin vacuum-deposited gold conductive layer. Very fast transient response and wide bandwidth are assured. With its high sensitivity (−44 dBm) and low distortion, the AT4031 meets the stringent dynamic range demands of today's storage and playback media.

Uniform, predictable performance

You'll find the AT4031 very easy to use. The uniform, predictable cardioid pattern translates into off-axis attenuation, not coloration. An integral second-order high-pass filter lets you roll off the bass at 12 dB per octave, when wind noise or room rumble must be controlled. An effective foam windscreen is also standard.

No-compromise performance

The AT4031 is intended for those professionals who will accept no compromise in performance, yet need a tough, reliable microphone. But, listen for yourself. Based on extensive field tests in studios and on stages around the country, we think you'll agree that what you hear is precisely what you need...the new AT4031 Studio Cardioid Capacitor microphone.

audio-technica®
1221 Commerce Drive, Stow, OH 44224
(216) 686-2600