MAY/JUNE 1990•A GPI PUBLICATION

EQ

THE CREATIVE RECORDING MAGAZINE

DAT

Unraveling The Tangled Story Of Digital Audio Tape

SPECIAL!

- •Arranging In The Studio
- •Chip Davis & David Grusin
- •PLUS! 24-Track Showdown

$3.95
Can. $4.25
U.K. £2.50

Display Until
6/30

0 262937 7

06 90

EQ.

VOLUME I, ISSUE #2 MAY/JUN 1990

THE

CREATIVE

RECORDING

MAGAZINE

26
COVER STORY

What's all the fuss about
DAT? Why isn't it widely
available? Will SCMS
lower the piracy poten-
tial? Will DAT be eclipsed
by recordable CDs? The
story of the little tape
that was too good.

From The Firing Line

WHEN *EQ* WAS STILL IN DEVELOPMENT last year, we spent a lot of time at the office talking about product reviews. We knew the quality of our reviews could make or break our credibility. Late one night, we hammered it out: The reviews would have to be comprehensive, fair, and tough. We would report the truth about a product—from our perspective—even if it meant offending a potential advertiser. We rolled up our sleeves and started testing.

As I write this, it has been three weeks since the first issue of *EQ* hit the stands, and three weeks since a few manufacturers hit the roof. One of them growled that we were off to a "terrible start" with his company. Another couldn't understand how we could say *anything* bad about her product (and we generally liked it). Someone else hollered through the phone that we didn't have a *right* to review his gear! Some folks out there are angry with *EQ*. Those have been hard phone calls.

They weren't the only calls, however. We're receiving an incredible amount of support for the approach we're taking to reviews, even from people whose gear we criticized. One manufacturer wanted greater details about our procedures, to help improve a product we tested. A major console maker confessed, "I fully expected to hate the monitor listening test, but I was wrong. I appreciate that you told the readers of the test's strengths *and* weaknesses." Observed a marketing director: "Once readers come to respect *EQ* as *the* source for reviews, it will serve us well when you review a product of ours that you really like."

We're not out to slam products, nor are we out to make enemies: The people who design, build, and market studio gear and instruments genuinely care about music and recording. Many devote themselves to the development of a single product for months or even years. Their hard work makes recording easier, better, and more fun for all of us. They make *EQ* possible. When we criticize something they've *birthed*, I don't blame them for getting upset. I'm grateful for the efforts put into any new product.

But you work hard, too. You have a right to know how a piece of gear performs, and how it might serve you in your craft—for better or worse. Ours is a competitive market, and anyone who's willing to sell you something should have enough *chutzpah* to welcome their gear being put to the test.

Our Lab Tests serve two purposes: They allow for more objective comparisons, and they may reveal deficiencies that don't crop up during hands-on tests. Each product receives a thorough workout in a real-world studio situation. Most see active session time, which puts a product's value in perspective. If we encounter problems with our test procedures, we either pull the review or tell you what those problems were. When the review is done, we fax a copy to the manufacturer. This is not an opportunity for them to "approve" the review, or influence our conclusions. Rather, it's a chance to check for technical inaccuracies, to point out oversights, and to fill in any blanks ("Just how much *is* that MIDI toaster option?"). We take their input, and if necessary, we go back to test again. And then we report the truth, as we see it.

In the end, *EQ*'s product reviews are not for the manufacturers. They're not for our advertisers. They're for our readers. I hope they serve you well.

Brent Hurtig

future watch

Hypermedia.
HYPErmedia?
HYPERmedia!

BY LINDA JACOBSON

HOWEVER YOU TYPE IT, hypermedia means different things to different people. What it means to you today may help determine the path your recording journey takes as we head for the year 2000.

Back in '65, computer visionary Ted Nelson observed that we think and talk in a vastly different manner from the way we read books and watch films or TV. So he proposed *hypertext:* non-sequential writing that branches. A hypertext document consists of text chunks connected by links that offer you different pathways, encouraging mental tangents, like a lively conversation does. You choose a reading path following your interests or stream of consciousness—sort of how you read magazines like *Spy* or *Mad*, where seemingly unconnected info chunks appear all over the pages.

Bring in other stuff you can mess with on a PC—audio, video, graphics—and you get *hypermedia*. Some call it *interactive multimedia.* These terms cropped up as computer buzzwords in '87 when Apple gave us HyperCard, which lets plain folk write Macintosh applications incorporating text, graphics, and audio. The Mac II lets HyperCard pal it up with motion and still video, and color, 3D, animated graphics. DOS, Amiga, NeXT, and other computer platforms now support hypermedia programming, too.

Hypermedia offers *random-access control.* You sure don't have that when watching TV—you can't control the →

I really enjoyed the first issue of *EQ*, especially the head-to-head listening test with near-field monitors ("Electric Near-Field Acid Test"). Working in a small jingle studio, they're a real necessity, and your article will help when I think about buying a new pair. I do have one small complaint, though. My issue of *EQ* arrived in a plastic wrapper. Haven't you people ever heard of biodegradable materials? Thanks for listening.

DEBBY LILIENTHAL
New York, NY
Thanks for your input. A small number of promo copies were mailed in "polybag" wrappers. This is a less-than-ideal practice, and we have asked our printer to look into alternative materials for use in future promotions.

Congratulations on the first issue. Looks great and we all wish you well. I just looked at the near-field monitor evaluation, and I appreciate your candid note ("Reader Beware") regarding the criteria for the evaluation. It was a subjective evaluation, yes, but still valid I think, because ultimately, the end-user's perception is all. Still, I have a few points:

I didn't notice any mention of power handling. Subjectively speaking, one thing the Yamaha NS10M-S does exceptionally well is handle power—continually reproduce the transients of soloed (raw, not sampled) bass drum and bass. You can learn a good deal about a system listening to CDs. The rest of what you need to know is revealed by listening to raw unprocessed sounds through the system. I would like to have seen this criterion added to the evaluation. In looking for a small system to take with me to the studio, it doesn't matter how "good" a system sounds if it won't get *loud* enough. There is nothing more frustrating than trying to get sounds on a tracking date and having those woofers hitting the stops. There aren't many small systems that can handle the demands of basic tracks.

Your terms "bass extension" and "treble extension" were unclear to me. Do these terms refer to how well the speaker reproduces bass and treble transients, or how much bass or treble the speaker produces? Your glossary says the former, while the JBL 4406 review implies the latter.

Also, the bar charts have the appearance of "the absolute," yet ten listeners is a very small sample.

Lastly, a speaker's performance is significantly dependent on the amp it's connected to. While an amp may be happy driving one set of speakers, it may not like the look of an-

---

other speaker's load. When I auditioned several amps with a given set of speakers, the differences in the low end and the punchiness of the system were remarkable.

PETER CHAIKIN
Yamaha Corporation of America
Buena Park, California
While we didn't have a specific category for loud listening, the test participants were encouraged to audition at a wide range of levels. It's also true that the choice of amp may be an influence (we used a Crown Micro-Tech 600LX, 135 watts per channel). As for "bass and treble extension," the participants were listening to the monitors' "abilities to reproduce low and high frequencies," a category that addressed both low- and high-transient information, and how much was reproduced. If we had been free of space considerations, a separation of these criteria would have been a better-detailed approach.

I really enjoyed reading your first issue. Congratulations! I've got one question, though, concerning the "Electric Near-Field Acid Test." I'd like to know what compact discs were used to test the various monitors, because the choice of program material would, in my opinion, help validate (or invalidate) the findings. I also want to know what you used because I'm always on the lookout for good source material, especially solo piano, to demo our own HD-1 and 833 studio monitors!

MARK JOHNSON
Meyer Sound Laboratories
Berkeley, California
We would like to have tested your very fine monitors, but their retail price didn't meet one

ILLUSTRATION: SCOT HALPIN

LINE IN/LINE OUT

EQ

EDITORIAL DIRECTOR
PHIL HOOD

EDITOR
BRENT HURTIG

MANAGING EDITOR
LINDA JACOBSON

ASSOCIATE EDITOR
JEFF BURGER

SENIOR EDITOR
ELISA WELCH MULVANEY

Contributor Spotlights

This month our cover story is written by San Francisco-based investigative reporter Cary Tennis, who also authored last issue's look at the home vs. pro studio controversy. Look for Cary's byline in upcoming issues as well.

Our Peter Asher interview is the work of Amy Ziffer. Amy is a graduate of Berklee College of Music (B.M. audio engineering), and a former editor of *Home & Studio Recording*. Currently she is a freelance writer and an "unwilling resident" of Los Angeles. •

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*, 20085 Stevens Creek, Cupertino, CA 95014.

Writing For *EQ*
If you'd like to write something for us, send a query letter with samples of your writing and some information about yourself to editor Brent Hurtig, *EQ*, 20085 Stevens Creek, Cupertino, CA 95014. 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 (for which you'll be rewarded if we use them) to managing editor Linda Jacobson, *EQ*, 20085 Stevens Creek, Cupertino, CA 95014.

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, *EQ*, 20085 Stevens Creek, Cupertino, CA 95014.

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, *EQ*, 20085 Stevens Creek, Cupertino, CA 95014. 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. In addition, we review new records and CD releases, as well as instructional videos, software, and books, so send record/video/book release information to "Reviews" c/o Linda Jacobson, *EQ*, 20085 Stevens Creek, Cupertino, CA 95014.

Permission & Reprint Rights
To request permission to use all or part of an article appearing in *EQ*, write to Peggy Clapham, 20085 Stevens Creek, Cupertino, CA 95014. For advertising reprint rights, send your request to our advertising director, David Sweet, at the same address.

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 (20085 Stevens Creek, Cupertino, CA 95014). Telephone 408-446-1105. Fax 408-446-1088. Office hours, 9-5:30 PST.

Overseas Advertising
Our advertising representative for the British Isles and Scandinavia is David Landsdowne. Contact him at Landsdowne Media Services, Ltd., Landsdowne House, 12 Church Street, Rickmansworth, Hertfordshire 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.

future watch continued

pace, or pause to reflect upon an idea. Picture a TV show about the blues; how long you watch any scene or listen to Muddy is determined by the transmitter. Sure, you can videotape it, and play Muddy's part over and over again, and wear out the tape. That's *some* control. You can even watch the Home Shopping Network. Some cable regions carry "videotex" and "teletext" systems that offer limited, searchable information to homes with touchtone phones, but you have to punch in numbers to get any info, and graphics are low-quality. How retro!

Enter optical technology. In the late '70s, MCA introduced the first consumer optical videodisc machine, using the new laser videodisc (LV) technology. Soon Pioneer, Sony, TEAC, and others built decks for the 12" videodisc platter. By 1979, companies like General Motors were playing videodiscs in showrooms and sales training offices. Then, in 1981, "levels of interactivity" were standardized to define the technologies used in interactive videodisc programs—an LV program is divided into digitally encoded frames, so videodisc players can select and tag sequences, though they really can't handle non-linear, alternative paths of access and display. For that they need help from computers.

Then came CD. The 5" compact disc uses a variant of LV technology to store up to 74 minutes of stereo digital audio. From this basic stock has poured an alphabet soup of storage/playback formats, some theoretically delicious: CD-V (20 minutes of audio, five of video); CD+MIDI (audio, graphics, and MIDI data); CD-G (audio with medium-res graphics); CD-ROM (650MB of text, graphics, and sound); CD-I (hours of sound, motion/still video, graphics, data, all under interactive computer control); and DVI. (Digital Video Interactive is for DOS and OS/2 users, and stores up to 72 minutes of video, graphics, and text.)

So the soup's here. Where's the beef? CD-V isn't good for much else except portable music videos. CD-G is big in Japan's sing-along bars, and sold as a music video mutant in record stores here, but has anyone on your block bought a CD-G decoder? CD-I would be great for scientific simulations, among myriad other things, but no hardware or software exists. (The →

The Wizard Of IT

This is the first of our Guest Editorials, which will be a regular feature of EQ. Our first Guest Editorialist writes in response to last issue's cover story, which examined the controversy in Los Angeles concerning professional recording studios and "illegal" and "unzoned" home studios that cater to outside clients.
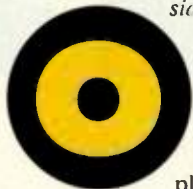
NCE UPON A TIME, long, long ago, when the world was new, people used to live and work in the same place. Some people worked at home, some nearby in the village. Children grew up knowing how their families worked, sharing troubles and joys, adventures and routine. People felt in control of their lives.

Then a great magic transformed the land. Called the "Industrial Revolution," it enticed the villagers away from their homes, brought them to the ever-growing cities with promises of excitement and comforts beyond imagining. Instead of working at home, the people learned to work in great factories, to mass-produce all the wondrous inventions of the wizard called Manufacturing Technology ("MT" for short).

As the wizard's magic grew in strength, some problems began to appear. The cities became more and more crowded. The air filled with fumes. The crowding produced increasing violence. Many people began to feel constrained and oppressed, no longer in control of their lives. The wizard's magic was a magic of increasing scale: Only the big could keep power—big companies, big government.

Then a new wizard appeared, called Information Technology, or "IT." IT grew at a prodigious rate. IT started giving power to the people. IT let them communicate with

---

Jack Nilles launched his career designing spacecraft for the U.S. Air Force and NASA; he then directed interdisciplinary research at the University of Southern California's Center for Futures Research. He coined the word telecommuting in 1973. Now, as a consultant with JALA Associates, Inc., Jack is developing telecommuting projects for the State of California, City of Los Angeles, and several BIG companies. His wife Laila Nilles produces classical records for Protone Records. And yes, they work at home.

each other, even see and hear each other at great distances. IT gave them new tools for thinking, for creating sounds, for creating visions, for creating worlds. In just a few score of years, IT's magic became the means by which most people were doing their work.[1]

But MT wasn't defeated by this. You see, MT had captured the hearts and minds of the people. MT made them think that they had to be where it's big in order to survive, in order to have all the goodies that MT's magic produced. They thought they had to *go* to work to go to work. They thought that work *had to be* separate from where they lived. The concept of BIGness had become the important thing to the people—and not what bigness actually did for them.

But not for everyone. Some people (the others thought they were crazy, or weird, or subversive) began to use IT's magic to restructure their lives. They found that they could work at home again, be with their fami-

CONTINUED ON PAGE 22

> "People thought they had to *go* to work to go to work."

BY JACK NILLES

There's no faster, easier or better way to record incredible stereo.

When you compare Crown's new Stereo Ambient Sampling System™ (SASS™) and a DAT recorder with traditional recording methods you'll discover there's no faster or better way to record natural, beautifully imaged stereo.

Lightweight, durable and extremely easy to set up, the SASS microphone is an exciting improvement in stereo recording. Combined with a DAT machine, it becomes a high-quality, no-compromise recording system that goes everywhere.

Crown's SASS 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.

The SASS is available in two versions: the SASS-P, with switchable battery or phantom power and Crown's finest studio-grade PZM®

capsules; or the SASS-B, which uses the famed Bruel & Kjaer 4003 and 4006 studio mics (not supplied).

Regardless of which you choose, 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 are—from sampling to electronic news gathering to remote recording of live events—you'll find Crown's SASS family the simple choice. See your Crown representative or call toll-free for information: 1-800-535-6289.

⌂ crown.

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

© Crown International, Inc.

future watch continued

powers-that-be have to figure out if it'll be sold in computer stores, stereo stores, or K-mart.) Now, CD-ROM is enjoying growing popularity as a manual and database delivery platform for health care, government, libraries, and higher education—especially since drives now go for under $1,000. (These drives run too slowly to be depended on for truly interactive hypermedia.) Some recording studios use CD-ROM to store their humongous sound libraries.

Today, Mac II and DOS-based systems can support graphics, text, and the user interface on the computer monitor, and, working simultaneously with LV or CD-ROM, can display moving images on a TV monitor—accommodating user hyperinteraction. Industry-wide design standards for interactive program producers do exist, and there is a growing number of intriguing applications. So instead of being media consumers, we can be media manipulators.

Current computer software programs allow us to create, edit, notate, and store the data we call music. But they fall short in the areas of archiving, annotation, analysis, and educational study, all of which invite a hyper approach. Considering the way audio data scarfs up disk sectors, it's impractical to rely on magnetic media storage alone for this. But by combining CD-ROM and mag media, possibilities open up. F'rinstance: The Voyager Co. (Santa Monica, CA) recently released the acclaimed *Symphony No. 9 Companion* ($99.95), an interactive exploration of Beethoven's masterpiece, for HyperCarded Macs with CD-ROM drives. An educator at San Francisco State University has developed an audio tutorial that incorporates CD-ROM, videodisc, digital keyboard sampler, and the Mac to teach audio production principles and techniques in an interactive format (presented at AES Montreux in March).

Right now, the best examples of fledgling hypermedia-for-the-masses come in HyperCard form. In fact, last week GM sent me a "Buick Dimensions" HyperCard stack that uses animated A/V sequences to show off their cars. It's neat, considering it's retail propanganda. The label says, "Make a copy and pass on to your friends." So, friends, if you'd like a copy, send a blank, Mac-formatted 800K floppy and SASE to Future Watch at *EQ*. See for yourself if herein lies the potential for a Golden Age of Information.•

Remote Possibilities

STANDARDS CAN be a beautiful thing. One example of a successful industry standard is the Electronic Industries Association's 19" rackmount specification. By now we take it for granted that almost every addition to a system will screw neatly into a rack. Another shining triumph, the MIDI standard, has settled down to serve as a reliable conduit of information between the products of different manufacturers. The proliferation of both rackmount and MIDI-equipped devices brings us to the focus of this column: the impending onslaught of remote control pandemonium.

Any user of household techno-wiz goodies is familiar with the problem. First a VCR is acquired, along with its proprietary remote. Then a CD player is added. The TV is upgraded to a high-powered, fully remoted model. A new tuner/receiver is plugged in. Soon a pile of infrared remotes is stacked on your lap, and Murphy's Law dictates that the one you need has been left in the other room.

Fortunately, a few intrepid souls figured out that a universal household remote could be produced. This device learns the infrared patterns of individual audio/video units and stores them in memory, so the pile of remotes can be reduced to a single one. There's still a key flaw in this arrangement, though: The remote's button functions change according to which unit they're addressing. Once you get beyond dealing with the controls of the first or second unit, you practically need a *Concentration* wizard to help you keep all the button functions sorted out.

So what's the connection between this household dilemma and the working professional? The increasing use of rackmounted, MIDI-controlled synths, samplers, and effects. In theory the concept is beautiful: Instead of endlessly replicating keyboards, use a single keyboard controller and rack up the rest. And everything works great—until you actually need to change a parameter on a rackmounted synth, or alter the wet/dry mix of an effect. Typically,

---

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

you'll find yourself slouching down close to floor level and developing lower back pains. There is an answer for us: the remote control. A healthy number of devices already offer this feature or provide it as an option. And it's a safe bet that even more manufacturers will go this route to solve the hard-to-program-in-the-rack puzzle.

Now the fun begins. The work space you recently organized and tidied by stuffing equipment into a rack soon becomes a clutter of incompatible remote controls made by various manufacturers. There is a better way: Let's create a pro audio/musical instrument control bus standard so that one universal remote can do the trick.

The type of universal remote used with VCRs and consumer electronics won't do, because infrared systems depend on line-of-sight operation. This often is not possible in a studio, and limits equipment installation and placement possibilities. So most likely a wired system—or something like Tascam's re-

Remote Control: The Next Industry Standard?

BY J.D. SHARP
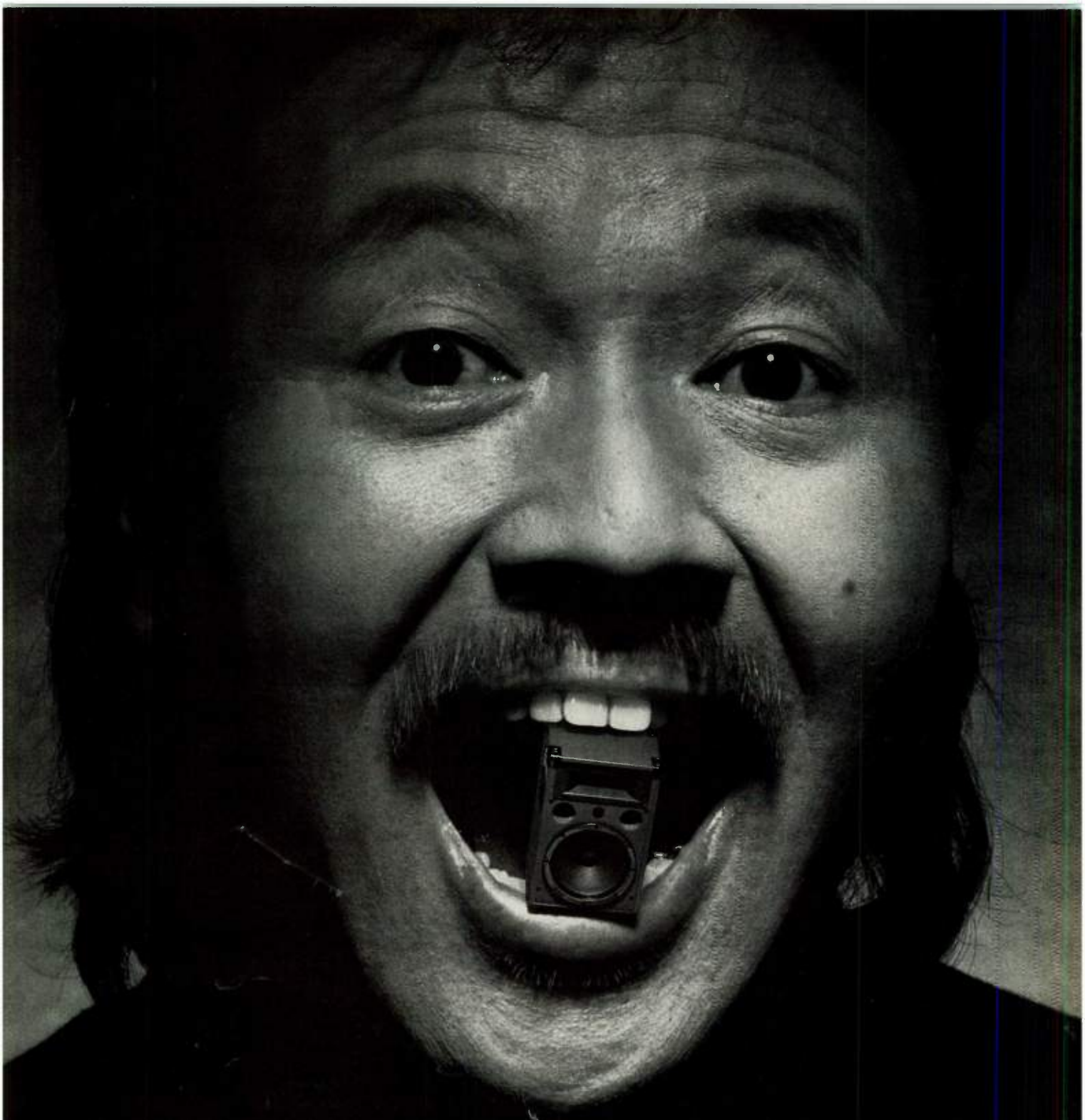
UPDATE

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

Compiled by Jeff Burger

Multiple Multi-Effects

REMEMBER WHEN reverbs were the size of radiators? When digital reverb and effects squeezed into a single rack space? And when they started doing more than one effect simultaneously? Well, folks, it's getting pretty wild. The **Zoom** 9002 ($479.00) packs 11 types of 16-bit effects into a palm-size package that weighs all of 280 grams! You get compression, tube-type distortion, EQ, pitch shift, phaser, flanger, chorus, two digital delays, and two reverbs. To top it off, you can have any six simultaneously. The unit comes with 40 presets, with room for 20 user programs. Connections are provided for headphones and mix output for recording.

Just in case you don't think Zoom is hunting for big game, their 9010 ($1,479.00) is a rackmount multi-processor with a choice of four completely independent configurations ranging from in-line to mixed to parallel. There are over 20 types of effects, and more can be added via ROM card. This unit has 128 factory presets and 60 user programs, all MIDI-selectable, of course. Frequency response is said to be 16Hz to 20kHz with over 90dB dynamic range.

ART ups the ante with the SGE Mach II ($749.00) with 12 simultaneous effects out of more than 70. Other features on this wonder include completely programmable parameters, two octaves of pitch transposing, two seconds of sampling, MIDI data monitor, compressor/limiter/expander, EQ, overdrive, Leslie, harmonic exciter...the list goes on.

If you want to put the pedal to the metal without a stack of Marshalls, **DigiTech** has combined all the toys in the MM-4 Metal Machine ($369.95). One of seven distortions can be routed to one of 15 delay/reverb effects, then processed with compression and 4-band EQ. It's not programmable (knobs, what a concept!). It's not MIDI. It just wails.

If you're using two or more effects units at a time, consider **Uptown Technologies'** Flash ($479.00), a veritable "Swiss Army knife" of MIDI-controlled audio routing. While it's tough to describe fully in a few words, some of its many uses include stereo switching (4x1 or 8x2 and 1x4 or 2x8), multiple-loop effects processing, MIDI-controlled multiple triggering for synths/tape decks, and MIDI on/off control over any AC device. •

Recordable CD

RECORDABLE CD technology is definitely here—at least for reference discs. The latest trend in mastering sends producers and artists home with a working master on an optical disc that they can pop into any CD player. No tapes. No clean rooms. Instant CD.

In the limelight these days are two systems that both incorporate combinations of equipment from different manufacturers. **Sonic Solutions** manufactures the Sonic System ($23,000 and up), a Macintosh-based system for digital editing, mixing, and CD pre-mastering. Its output is encoded and recorded onto write-once optical media with CD Maker ($25,000) from Start Lab, a joint venture between Sony and Taiyo Yuden. Up to 31 additional recorders can be added for $15,000 each. The blank CDs cost about $40.

The older CDR 90 ($49,500.00) comes from **Gotham Audio**, and is installed in several of the country's top mastering houses. This is another integrated system, combining Fuji's write-once disks, Yamaha's YPE-101 encoder and YPR-101 optical recorder, Harmonia Mundi's interfaces, and an IBM PC/AT-compatible computer. Its optical disks cost between $80 and $90.

At $40 to $90 per disk—for just the storage media—these systems won't replace current CD manufacturing techniques, but will enjoy widespread use for short archival, reference, and broadcasting runs. One day soon you may be able to leave your local studio with a master CD in hand, costing about $350—expensive when compared to a DAT, but a bargain when compared with the average $900 glass master. •

CyberArts Conference

GET READY FOR CYBERARTS INTERNATIONAL—a professional showcase of new tools, techniques, and training for artists in interactive multimedia, co-hosted by *EQ* and *Keyboard* magazines. Some highlights will include performances and presentations encompassing virtual reality, interactive video, computer-modeled holography, DSP, computer animation, sound effects/film scoring, new instrument design, and...well, just about anything else cool you can think of that has to do with technology in the arts! Four-day admission is $450.00 ($395.00 if paid before June 30). The conference will be held September 6-9, 1990, at the Los Angeles Biltmore Hotel. For information contact: CyberArts, 20085 Stevens Creek Blvd., Cupertino, CA 95014; (408) 446-1105. •

Rit Shoots Jazz Video

WHEN LEE RITENHOUR gets a few of his close friends together, great music is inevitable. "Rit" recently staged a sophisticated video shoot with a virtual who's who of jazz flowing on and off the stage, including Bob James, Harvey Mason, Anthony Jackson, Ernie Watts, Tuck & Patti, Paulinho Da Costa, Joao Bosco, and even Steve Lukather. A Westwood One remote truck with 48-track digital audio was recruited to help pull off the seven-camera shoot at the Coconut Grove club in L.A.'s old Ambassador Hotel.

The star-studded event was staged to create not one, but two different videos for the Tokyo-based VideoArts Japan. The video also will be available in the States, but the company admits that U.S. sales alone probably wouldn't cover the $100,000 production costs. Fortunately for Ritenhour and friends, the Land Of The Rising Sun is almost as hungry for American jazz videos on laserdisc as it is for American real estate. •

Digital Audio For Personal Computers

DIRECT-TO-HARD disk recording options are increasingly available for most popular computers, largely due to the ever-increasing speed of today's hard drives. These new systems typically need a drive with 28ms access time and a controller with 1:1 interleave. (As a rule of thumb, storage capacity is 10MB per minute of stereo material at a 44.1kHz sampling rate, with additional overhead required for the operating system and software.)

For Apple's Mac, **Digidesign** and **Opcode** have collaborated to combine two popular products, SoundTools and Vision. The result—Digital Audio Vision—offers an integrated environment that adds the power of 2-track digital recording to the professional MIDI sequencing environment. Imagine cutting and pasting MIDI *and* digital audio passages, with one fell swoop of the mouse! This especially is great news for composers/producers who do most of their work with virtual tracks, employing tape only for voice or a

live solo. SoundTools costs $3,250.00 and the enhanced Vision front-end goes for $695.00.

Digital Dynamics' ProDisk-464 is the first Mac-based system we know that is expandable from four to 64 tracks. This is accomplished by racking up an additional hard drive for each 4-track increment. A full-blown system can handle up to eight track-hours.

Digidesign also has released SoundTools ($2,995.00) for the Atari Mega ST, allowing digital audio to be recorded directly to any Atari hard drive. Consisting of an external A-to-D/D-to-A converter and accompanying software, the package features non-destructive playlist editing, digital equalization/mixing/merging, and time compression/expansion—all using the familiar mouse-based cut/copy/paste editing techniques. Options include time code synchronization via a SMPTE-to-MIDI converter, and digital interface for AES/EBU- and S/PDIF-protocol data transfers.

Lately the most digital audio action has been occurring in the ·IBM PC/clone arena. **Turtle Beach** released the 56K System for the IBM AT or 386. The interesting thing

RECORDING CONSOLE design is maturing at many price points. Manufacturers focus much attention on accommodating more tape tracks and virtual tracks in more compact designs.

At the high-end, the Amek Mozart is available in 40-, 56-, or 80-input frames ($97K to $344K). All models have 32 output buses for recorders and 12 stereo returns, with up to 16 aux buses available, depending on choice of input module. Automation software co-developed by Steinberg works in conjunction with a trackball built into the console and a color monitor: Movements of 96 faders and 15 switches per input can be recorded live or off-line into the software. All automation functions can be cued against SMPTE time code.

Tascam has announced the M-3500 series, designed for the multitrack-plus-MIDI project studio. The M-3500/24 ($6,999.00) has 24 channels, while the M-3500/32

here is that the 56K will use the A-to-D and D-to-A converters on a DAT machine, which cuts down on the cost ($2,689.00); the 56K Analog/Digital Converter is a $995.00 option on this system! Considering the cost of mail-order clones, this could be the least expensive system around. The company also will configure a turnkey system on request.

Eltekon's MicroSound provides disk-based recording, 2-channel ($3,695.00) or 4-channel ($4,995.00), for any 386-based machine. More channels can be accessed simultaneously by synchronizing additional processors. •

($7,999.00) has 32—the additional eight channels with optional stereo input. The modified in-line monitoring design provides a short fader, two stereo effects sends, pan, and mute for tape monitoring while tracking; the FLIP switch reverses the inputs on mixdown, so tape tracks take advantage of 4-stage EQ and a long fader, while virtual tracks come in via the monitoring section.

Another modified in-line offering, Maxcon, comes from CAD. This console is extremely compact, starting out with 16 channels in a rack ($6,599.00), expanding in 4-channel increments to a 96-channel console ($34,199.00). The compact nature owes to two basic design concepts. One: each input not only crams in six aux sends, eight subgroup assigns, 4-band (three sweep) EQ, and a tape return pot, but the 10-segment LED meter is placed parallel to the fader to conserve space. (As an additional trick, the tape monitor alternately can be used

MANUFACTURER LISTING

Amek, Regent Trading Estate, Oldfield Rd., Salford, M5 4SX, England, (016) 834-6747 [U.S. distribution: AMEK/TAC, 10815 Burbank Blvd., North Hollywood, CA 91601, (818) 508-9788]; **Anatek**, 400 Brookbank Ave., North Vancouver, BC, Canada V7J 1G9, (604) 980-7061); **ART**, 215 Tremont St., Rochester, NY 14608, (716) 436-2720; **CAD**, P.O. Box 120, Conneaut, OH 44030, (216) 593-5395; **Digidesign**, 1360 Willow Rd., Menlo Park, CA 94025, (415) 327-8811; **Digital Dynamics**, 270-02 E. Pulaski Rd., Greenlawn, NY 11740, (516) 271-5600; **Digital Music**, 5312-J Derry Ave., Agoura Hills, CA 91301, (818) 991-3881; **DigiTech**, 5639 South Riley Lane, Salt Lake City, UT 84107, (801) 268-8400; **Eltekon**, 37493 Schoolcraft Rd., Livonia, MI 48150, (313) 462-3155; **Gotham Audio**, 1790 Broadway, New York, NY 10019, (212) 765-3410; **Intone**, 14913 Praire Ave., Lawndale, CA 90260, (213) 644-2508; **J.L. Cooper**, 13478 Beach Ave., Marina Del Rey, CA

as a seventh stereo aux send!) Two: the narrow master module stacks the eight submasters vertically in pairs above the stereo master faders; even after adding the aux returns and other controls, the entire master module is only the width of two channel modules.

The Soundtracs Prism also owes its compact size to in-line design. This 24-bus console is offered in 32-input (around $30K) and 48-input (around $45K) versions. The Prism also includes mute automation of channels and auxiliaries: Any MIDI sequencer can control these functions via simple NOTE ON/NOTE OFF commands. (In last issue's look at MIDI automation, we didn't mention that Soundtracs had one of the first consoles to support this feature—the now-discontinued CM4400.)

All these consoles take today's trends into consideration: More and more producers are leaning toward using sequencers to play a stack of MIDI devices during the mix. That means more inputs. And that means more music. •

More MIDI Channels Please

WHILE NOBODY'S SHOUTING about MIDI 2.0, various companies are tackling the problems of advanced MIDI networking. Given today's multi-timbral instruments, MIDI-controlled effects, and MIDI-based mixing, one big problem is the need to address more than 16 channels worth of discrete data.

The most ambitious effort comes from **Lone Wolf**, makers of MIDITap and the new FiberLink. The system concept is a local area network capable of handling a half-million (yes, we said a *half-million*) MIDI channels, digital audio, digital video, SMPTE time code, and just about any other digital data over fiber optic cables. Each MIDITap ($2,500.00) supports four discrete MIDI inputs and outputs, for a total of 64 channels per unit; multiple MIDITaps can be linked together via fiber optic cables. Rather than being identified by a simple MIDI channel number, a MIDI device is addressed as one of 16 MIDI channels on one of four ports on a given MIDITap. FiberLink allows the system to send MIDI data over fiber optic cables up to 2.5 kilometers in length!

Mark Of The Unicorn has released MIDI Time Piece ($495.00), a MIDI interface/patchbay/SMPTE-to-MIDI converter for the Apple Macintosh, with eight MIDI inputs and outputs. Each port

acts as a separately addressable 16-channel bus for a total of 128 channels. A network port allows a second MIDI Time Piece to be integrated for a 16x16 function with 256 channels. All that still only takes a single Mac port; a separate 16x16 setup can be connected to the second port using up to 1,000' of cable. While currently the product's full potential can be realized only with Performer Version 3.4, other manufacturers no doubt will start supporting this unit; in the meantime, a desk accessory will configure merging, routing, channelizing, and event filtering as if the unit were a dual 16-channel interface.

MIDIBuddy ($1,495.00) from **Acme Digital** offers ten discrete inputs and outputs for a total of 160 virtual channels. This product also supports a proprietary fiber-optic expansion port option. MIDI data processing is included for channel and control remapping, programmable delays, filtering, transposition, inversion, and velocity limiting. Setup changes can be triggered by just about any kind of MIDI event. MIDIBuddy is distributed by Eltekon.

And what about IBM PS/2s and portable PCs? **Key Electronics** not only makes an external interface for these machines, but their MIDIATOR MS-103 ($179.95) supports 48 MIDI output channels via three discrete jacks.

Aiming above and beyond Apple- and IBM-type ma-chines, **Uninet** now offers MIDI support for Sun, DEC, and VAX workstations ($900 to $3,600); appropriate drivers allow MS-DOS and Mac music software to run in windows.

Variations on the basic programmable MIDI patchbay theme are multiplying like rabbits: **KMX** has announced their 8x8 MIDI Patch Bay ($249.00) with 30 programmable combinations that can be recalled via MIDI PROGRAM CHANGE or SYSEX commands. **J.L. Cooper** has added features and cut costs with the MSB Plus Rev 2 ($389.00), an 8x8 with lots of bells and whistles. For bigger applications, the same manufacturer is shipping Synapse ($1,195.00) with 16 inputs and 20 outputs. **Intone**'s MIDI Maestro ($995.00) is simultaneously a 6-in/8-out MIDI patchbay, and a 16x16 audio patchbay.

You don't necessarily have to buy a MIDI patchbay to route appropriate MIDI devices back to a computer for SysEx handshaking, however. Some new entries that do the trick are the **Anatek** Studio Merge ($399.99), which merges eight MIDI signals to one, and The Funnel ($79.00) from **Digital Music**—an active device that senses which of its inputs is sending MIDI data and routes it to the single output. •

EQ is the brand new magazine that covers the real-life world of today's recording studio.

From recording techniques to studio and career management, from programming and performing to hands-on and how-to, *EQ* is for engineers, producers, and musicians who are serious about recording. Each and every bi-monthly issue is packed with pages and pages of . . .

● Hard-hitting reviews of today's hottest new recording gear, MIDI equipment, software, and more.

● Fascinating visits to artists' and producers' studios, as well as commercial and home facilities all over the U.S. and the world.

● Audio-For-Video - you'll learn the inside workings of music video, film, and television sound production.

● Workshops and clinics, on everything from Arranging In The Studio to Digital Recording, will give you the competitive edge.

And that's not all. You get advice from leading engineers, producers, and artists as they talk about their latest projects. *EQ* will also help you get a real-world perspective on how recording professionals make their moves into the job they want, and how independent "wizards of the board" build their own studios into profit centers. Plus, monthly columns covering basic recording tips, computers, tax and legal advice, studio maintenance, and much more.

Take advantage of the special CHARTER SUBSCRIBER rate that saves you up to 40% off the regular rate. Be part of the only magazine for serious creative recording.

EQ
BECOME A CHARTER SUBSCRIBER TODAY!

PREMIER ISSUE

EQ

THE CREATIVE RECORDING MAGAZINE

STUDIO WARS

REELS OF FIRE: SPARKS FLY AS 'PRO' AND 'HOME' RECORDING WORLDS COLLIDE
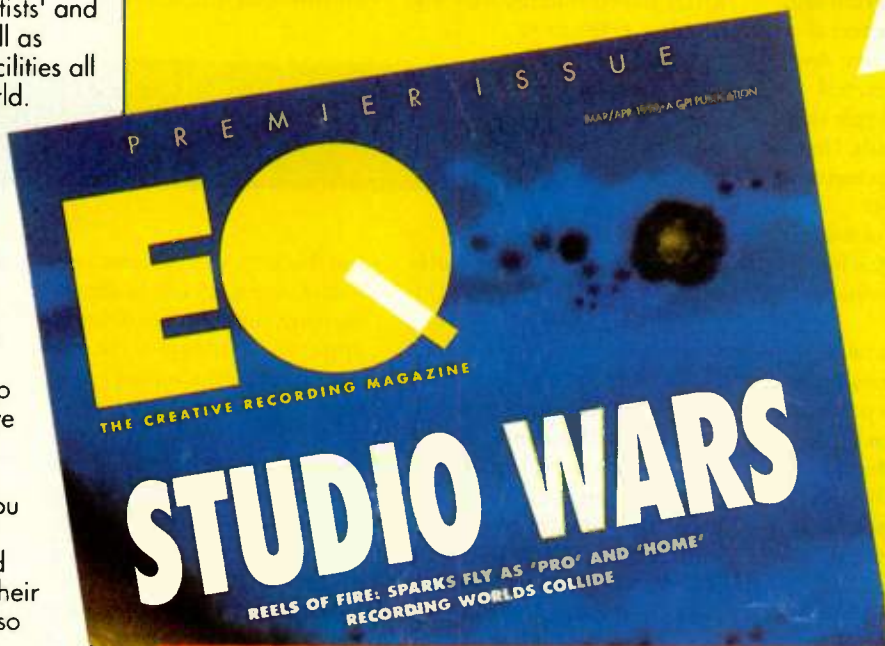
YES! Enroll me as a CHARTER SUBSCRIBER to *EQ*.
- ❏ **$19.97** for 1 year (6 bi-monthly issues), **29%** off the basic rate! *
- ❏ **$33.97** for 2 years (12 bi-monthly issues), **39%** off the basic rate! *
- ❏ **$49.97** for 3 years (18 bi-monthly issues), **40%** off the basic rate! *

Send to: EQ, P.O. Box 57295, Boulder, CO 80322-7295
❏ Bill Me ❏ Check/Money Order ❏ VISA ❏ MasterCard ❏ American Express

Card # _____ — _____ — _____ — _____

Expiration Date _____ Signature _____

Name _____

Address _____

City _____

State/Prov. _____ Zip/P.C. _____

Prices good in U.S. only. Canadian and international surface mail subscriptions add U.S. $5 per year. International air mail subscriptions add U.S. $20 per year. All non-U.S. orders payable in U.S. dollars by VISA, MasterCard, American Express, or International Money Order only. Allow 4-6 weeks for delivery of first issue. *Basic rate 1 year (6 bi-monthly issues) for $27.95.

50EE9

Dave Grusin

Grammy Winner Goes For Scoring Without Rules

BY NICK ARMINGTON & LARS LOFAS

'M THE MOST undisciplined person I know," laughs composer, producer, and record company owner Dave Grusin, when asked about his tactics for completing film scores in unmercifully short timespans. "I react best to the panic of a deadline. My creativity comes in bursts, from some source that I don't pretend to understand. So I just hammer away at things, and hope that the best stuff is what we end up with, not what gets abandoned. But sometimes it's hard to know."

What is known is that Grusin's method works. During a career that spans over 30 years, the 55-year-old Grusin has earned wide respect for his composing and arranging, numerous recordings, and the success of GRP Records, the record company he founded in 1976 with partner Larry

Rosen. (GRP recently was sold to MCA Records.) His film music credits, including *Tootsie, On Golden Pond*, 1988's Oscar-winning *The Milagro Beanfield War*, and *The Fabulous Baker Boys* (for which he recently won a Grammy), place him at the top of his profession.

Grusin got into arranging right from the beginning of his career. After earning a music degree from the University of Colorado, he relocated to New York City, intending to continue his studies. Instead, he hit the road playing piano with crooner Andy Williams. When Williams got a break hosting a weekly TV variety show in Los Angeles, Grusin signed on as the show's music director. "That job was a tough assignment," he recalls. "Writing a new show every week, I really learned about arranging on the fly. Fear can be the best teacher."

Despite the Hollywood credentials, the soft-spoken Grusin prefers to spend most of his time living and writing in New Mexico and Montana. He works on an average of two film scores a year, while also releasing solo albums and overseeing his label's diverse roster of musicians.

Grusin eschews the complex studios that many of his film-scoring colleagues use for their work, preferring to compose on a simplified MIDI system unceremoniously arranged in the smallest of three bedrooms at his Santa Fe home. The setup is based on an Atari 1040ST computer running Hybrid Arts' SMPTEtrak sequencing software. It includes a Yamaha DX7-IIFD, Korg M1, Roland MKS-20 piano module, Roland MKS-80 Super Jupiter synth module, a rackmount 16x4x2 Hill Multimix mixing board, and a DAT recorder by JVC.

"I deliberately don't have a full-scale recording studio at home. With electronics, you're constantly faced with the choice of either writing or just playing with the toys. It's hard to do both. Besides, my habit was always to work things out in my head, not on a keyboard. I've come to computer technology rather late, and have had to learn it from the back door in. But I love working with it now."

His compositional style may be low-tech, but the computer has proven an effective productivity tool for Grusin. "In the old days, you'd never know how something would sound until you got to the scoring stage with a full orchestra," he says. "Now, when I'm working with the VHS workprint of a film, I can at least put strings and piano sounds together, and use the instrumental timbres that actually will be played when we

Chip Davis

The Man Behind
Mannheim
Steamroller Makes
Arranging Look
Like Magic

**BY DEBORAH
STARR SEIBEL**

Also helping him clear the mental canvas of extraneous thoughts and worries is a fuzzy teddy bear. "My wife gave it to me six years ago when I went to record with the London Symphony Orchestra and I was really nervous. When I look at it, I know that someone is caring about how I'm doing."

But it's the magic wand with its cut crystal handle that helps clear Davis' path to the creative process and allows the artist to begin filling his musical canvas. "I get up really early when I'm composing," says Davis, "and I look through the handle at the moon." Calm, fearless, and inspired by the universe, Davis can begin the work of composing and arranging.

Despite his meditative creative method, Davis also is one of the most hard-nosed, cost-efficient producers in the business. He has been known to take a song from composition to computer print-out in less than two hours. But the studio in his basement—for all its 48 digital channels and MIDI bells and whistles—is just a highly sophisticated packaging tool for a single, all-important concept: the song.

"That's the most important thing," says Davis, who agrees with the notion that some music arrangers may be distracted, even confused, by the number of modern musical and technical options at their disposal. "I come from the standpoint that less is usually more.

"Take a song like 'Deck The Halls,'" suggests Davis, referring to a cut on Mannheim Steamroller's phenomenally successful 1985 holiday album, *A Mannheim Steamroller Christmas.* "The basic topic of that song is fun. You're not going to want an arrangement in a minor tone, or one that uses a lot of dark instruments. You try to follow what the song *means.*"

On a practical level, following the song's meaning for Davis involves establishing the chords, bass line, and melody to give the piece what he calls a "black and white framework," or solid foundation. "That basic framework should sound good," Davis says, "covering rhythm, tempo, and the brightness or darkness of the whole piece."

Then the "coloring" process begins, with Davis injecting what he calls the "textures and surfaces" of the music. "That's when I start adding strings," he notes, "and other instruments to enhance the mood and convey the

N THE MIDST OF ALL the electronic wizardry—a mind-numbing array of 20th-century synthesizers, samplers, and digital and computer technology—it comes as a jolt to know that one highly successful composer/arranger/producer/performer *never* enters the studio without his candle, teddy bear, and crystal wand.

"I have these good-luck things," says Chip Davis, the Omaha-based creative force behind classical-rock hybrid Mannheim Steamroller and the thematic "Fresh Aire" series of epic recordings known to new age enthusiasts and audiophiles everywhere. "I'm kind of superstitious." To this prolific composer, a simple candle offers a means of clearing the mind. "Flame has always mystified mankind," says Davis. "So when I have to totally compose something from scratch, I stare at it until my mind is a blank."

CyberArts™

INTERNATIONAL

A professional conference showcasing emerging Interactive and Multimedia technologies and their applications in the arts, **CyberArts** is a unique meeting ground for musicians, animators, videographers, directors and producers, and hardware and software designers working on:

Computer Animation	Algorithmic Art
Interactive Video	Multimedia Systems Authoring
Audio for Video & Film	Aesthetics
Virtual Reality	Laser Interactivity
Innovations in User Interface Design	Interactive Multimedia Theatre Production
Holophonics	
MIDI LANs in Studio & Stage Design	Holography
	Digital Sound & Image Processing
New Synthesis Techniques	and more

SEPTEMBER 6-9, 1990
LOS ANGELES BILTMORE HOTEL

PRESENTED BY
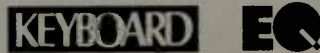
KEYBOARD　**EQ**

CyberArts International will provide you with an unprecendented forum for cross-fertilizing ideas, learning about the latest tools, and making valuable contacts for future projects.

KEYBOARD and *EQ* are publications of the GPI Group, a division of Miller Freeman Publications. **CyberArts International** produced and managed by Miller Freeman Expositions, San Francisco Group, 500 Howard Street, San Francisco, CA 94105. 415.995.2471 FAX 415.995.2494.

❑ Please register me for **CyberArts International.**
I wish to attend:
❑ Full conference. $450
($395 if received by 6/30)
Single day $165 (select day) ❑ 9/7 ❑ 9/8 ❑ 9/9
Send further information on ❑ attending ❑ exhibiting/sponsoring

Name _____ _____

Organization _____ _____

Address _____

City _____ State/Province, _____

Postal Code _____ Country _____

Phone _____ FAX _____

❑ VISA/MC/AMEX# _____ Expires _____

❑ Check/Money order
Payable to: **CyberArts International**, 500 Howard Street, San Francisco, CA 94105

EQ1-646

DAVID GRUSIN

record the score. I like to use the technology as a guide. I think of it as a really involved click track."

Another element of Grusin's success is his knack for choosing the right musicians to play on his projects, and leaving them plenty of room for improvisation. He explains, "In the pursuit of a finished product, you sometimes run into detours more interesting than your original idea. So when I work in the studio, there really aren't any rules. The whole process of letting everyone hear your arrangements and getting other people's feedback is really the way the music gets formed. No matter how concrete my original idea might be, I'm usually more excited after hearing other people's input."

This freeform production style emerges on Grusin's latest digital offering, *Migration* (on GRP), which he recorded with drummers Harvey Mason and Omar Hakim, bassists Marcus Miller and Abraham Laboriel, guitarist Carlos Rios, sax whiz Branford Marsalis, flugelhornist Hugh Masekela, and percussionist Mike Fisher. The album's rhythmic feel, Grusin notes, is derived from the improvisational nature of

"If there's anything I've learned from producing over the years, it's a willingness to let everybody in the studio find their way in the most musical way they know. If you've done your job calling in the right people, you're only going to benefit from giving them their say, at least until a tune is pretty well down. At that moment, you have to stop being a good guy, a fellow musician, and instead start listening to it in an objective way, analyzing how it would sound as a finished piece." •

its recording sessions.

"I would take each demo that had been done at home and select one element which became the core of whatever idea I was trying to get across. Usually it was not a melody, but a rhythm pattern played on the Yamaha RX-5 drum machine. I had worked on the rhythmic sounds and developed the tonality of the patterns to an extent where I knew I wanted them to be in the tracks, and that's what we would end up playing to as a guide."

Finding the right sonic textures for his arrangements, Grusin admits, calls for a solid dose of instinct rather than elaborate

programming. "The song tells you what to do, if you listen to it. I seldom think of a melody for something without attaching it to a sound."

His open-minded improvisational approach works equally well when making compositional choices for film scores. "I don't really have a philosophy of putting music to picture because it wouldn't hold up from one picture to the next. It's not so much what a film is about, or whether it's a love scene or chase scene; rather, it's largely what the film looks like in the cinematography and the editing."

Despite his enviable track record, Grusin still works hard on his scores, and claims the work, and the deadlines, are as challenging as ever: "Film scoring is still really hard for me. I kept thinking it would get easier as time went on, but it hasn't happened. If the style of a film hits me a certain way the first time I see it, that makes creating the music a lot easier. If you don't see it right away, it becomes real tough, but I guess that's the reason I do it. I can't imagine finding formulas for this stuff. Every picture is different, so every score should be different. The biggest trick is to try not to do the same thing every time!" •

CHIP DAVIS

feeling. For me, it's usually a building and refining process."

Perhaps surprisingly, Davis—whose recordings have a reputation for an astonishing array of natural and environmental sounds—is cautious when it comes to sound effects layering. "You have to be careful about adding things," he advises. "In the old way of producing, you get committed. But with the new equipment, you can do the process and then play with it, try different things and change your mind. Ultimately, I let my ear be my guide. You don't want to make 'Silent Night' sound like the score to *Ben Hur*."

Davis depends on his 18 synthesizers and samplers to get the right mix of sounds. But his favorite is the K1000 Kurzweil (he has four of these rackmount modules), a synthesizer capable of producing some 225 distinct sounds. He says, "I really enjoy its stereo imaging, the way a sound can move, like echo delayed from one speaker to another. It helps me define [arranging] space. It's actually changing the way I'm writing."

The rackmount Kurzweil 1000 SX sound module is another Davis favorite for its ac-

Producer's Tip:

"If you're painting a picture, you have to know the guidelines—the size of the frame, the kind of paint. It's the same with arranging. When you're writing a song, always write to a click track or metronome, because when you're beginning the foundation of your piece, it needs an inherent rhythm. Why? Because human beings need rhythm. If you speed it up or slow it down, it confuses people and they don't get it."•

curate simulation of orchestral strings. "It's close enough to the real thing to let me hear how it'll sound. Because for the final track, I always use real strings."

Davis moves from the arranging process to committing the song to paper with the help of Mark Of The Unicorn's Composer notation software, running on his Macintosh SE/30 (with a 40MB drive). That sets up the basic pattern. Then the notes are quantized—corrected for minor timing inaccuracies—with the help of Mark Of The Unicorn's Performer sequencing software. Only then is the song ready to be printed on paper.

For everything except the final song assembly, which involves live performance,

Davis uses all-digital technology so he can process multiple soundtracks without losing generational quality and adding extraneous noise. Indeed, his mixing system consists of six 8-channel Yamaha DMP7 digital mixer/signal processors, tied together to provide 48-channel capability, with a 192-point patchbay.

Only at the end of the process does Davis switch to analog tape, when he goes to Sound Recorders, a commercial recording facility in Omaha. He brings a DAT recording of his homework for reference. There, some synthetic parts are kept—synced to the studio's 24-track MCI JH-24 recorder—but many parts are replaced or are doubled with acoustic instruments.

For all his high-tech savvy and multitude of options, Davis is a composer who knows when to quit. "Eighty percent of the time we keep the first take," he said. "You can tweak it and tweak it and milk all the feeling right out of it. I don't like rearranging my own work. I like to move forward through time." •

lies, spend time in the villages where they lived, *and* support themselves as well as when they were with the BIG groups. They found that IT let them regain control of their lives, to learn, to create, to expand. IT started quietly at first, the faint few whispers of the incoming tide of change. A person here and there working for the BIGs started to work at home some of the time. A few used the growing power of IT to start their own businesses at home. As IT's magic grew, so did the number of people who discovered the old ways with the new tools. *They* began to build new home lives. IT grew. The number of "homies" (people who work at home) grew. To millions.

Then one of those homies crowed too loudly. The BIGs became alarmed. They said, "You're trying to destroy THE WAY! You're being UNFAIR! You're not playing by THE RULES! You're trying to change THE GAME!" They attacked the homie like ratters a rat. You see, they didn't realize that IT had made the rules obsolete, enabled the ways of work to be restructured.

But the tide of change kept coming in. The smarter organizations adapted the rules to the new ways. The rest of the world

started to use IT's magic to make new nows and new tomorrows. Governments fell. The world became more closely interconnected. The homies began to be in the majority. The cities began to be liveable once more.

And then—they all went down to the seashore and lived happily ever after, right? What's a magazine like this doing with such fairy tales? Let's get serious!

Information technology is growing in power (transistors per microchip, data capacity of fiber optics) per buck at about 28 percent a year. IT's been doing that for over 20 years. Think about it. That works out to a ten-times cost reduction about every seven years[2] for the information tools to do a growing number of information jobs. IT'll keep doing that into the 21st century.

IT is the basis for the revolution in recording of all sorts: audio, video, written communications. IT's magic is literally giving power to the people—especially professionals. The recording industry is but one of the areas where this is happening. Right now a low-budget home studio can produce high-quality digital audio master tapes. By 1995 or so, home studios could be

Then a new wizard appeared, called Information Technology, or IT. IT grew at a prodigious rate.

producing digital video master tapes for high-definition television, with computer-generated stereo (sound and video), fractal backgrounds, and animation. On top of that, fiber optics and communications satellites will be able to interlink production studios and customers economically, regardless of where they are. (Many BIG labels today are rumored to do their recording in Europe, mastering in the U.S., and CD production in Japan.) There's no telling what country that home studio will be in!

Yet BIG *can* be beautiful. For some things, like collecting massive resources to do well-understood or very complicated procedures, or quickly producing large quantities of stuff, BIG can't be beat. For others, like artistic creativity, there may be a better way. Starting at home.

Here comes the tide. •

[1] The information sector of the economy first supplanted manufacturing as the dominant sector in 1954 (about 4 P.M. on August 15, as I recall).

[2] The equation for this is the standard compounding equation: Total Price Reduction over N

years [TPR] = (1 - Annual Price Reduction $[APR])^N$. If the APR is 0.28 and N = 7, then TPR = $(1 - .28)^7 \approx 0.1$. The reason the rate is 28%(approx.) is an empirical law of semiconductor manufacture, called Moore's Law, Version II, which states: The number of logic elements on an average manufactured microelectronic chip will double about every two years. Version I of Moore's Law held true from about 1964 to 1974, and had an annual doubling time—which produces an order of magnitude change in about 3.4 years. My analysis of the rate of innovation in the semiconductor industry, including what's now in the labs, is that Version II will continue to be valid until about 2007.

SHARP ANGLE

mote for their rackmount CD player, which can be used wired *or* wireless—is needed.

Some might logically ask at this point why a MIDI controller like a Cooper Fadermaster or a Lexicon MRC-1 couldn't be used for this purpose. The missing element is an alphanumeric display above each button or set of buttons that can change along with the functions of the button, as well as an automatic recognition system between the remote and the device being controlled. (The MRC-1 does have a display, but it needs programming.)

In an ideal universal remote for professional audio, the name of the device being controlled would be displayed at the top of the window. A series of "soft" (programmable) keys would have "label" windows above them, and, for each device in the studio, the keys would be re-labelled as needed. A PAGE button would allow switching to different sets of functions, so the remote box wouldn't have to be immense to accommodate complex devices. LED or LCD meters would be nice, to display input or output levels when appropriate. It is questionable whether MIDI is the ideal format for a dedicated control bus; perhaps a variant called Control MIDI could be developed for this purpose. Anyway, the limited capacity of the MIDI bus already is overtaxed in many systems; more controller data running down the same lines hardly is needed.

Although this may sound like an expensive proposition, it should *reduce* the cost of rackmounted devices. Manufacturers could fabricate devices with blank front panels (except, of course, for the attractive screening of logos in day-glow colors),

Let's create a pro audio instrument control bus standard so that one universal remote can do the trick.

eliminating the incredible redundancy of switches and displays contained in a typical rack. Design energy could go into producing the nicest, most advanced remote-control panel.

The musical instrument industry miraculously pulled it together and cleaned up its act when it came to interconnection of synthesizers and effects. It's not too late to avoid creeping remote-control sprawl if action is taken soon. Although this cooperation is only remotely possible, it is immensely desirable. Otherwise, stock up on Dymo labeling tape and leave open a large area of your work space to contain an ever-growing pile of remote controls. •

IT'S ABOUT TIME

The Quest For Realism

BY NORM WEINBERG
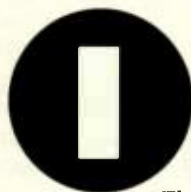
IN THE NEVER-ENDING QUEST for more realistic and "human"-sounding drum programming, many folks have abandoned their drum machines in favor of software and samplers. Why? The sequencer lurking inside your drum machine can't hold a candle to powerful high-end sequencing software. Likewise, samplers offer many more alternatives for altering a sound's timbre than most drum machines do.

In the last issue, we explored a few techniques to make your drum machine sound more natural: programming with pads, working in longer patterns, using a wider dynamic range, and recording in high-resolution mode. This time we're going to investigate a few additional tricks that can be applied when programming your drum tracks on a computer and using a sampler for a sound source.

Perfecting Imperfection. Let's say you've just recorded a hi-hat track without quantization. Most of the attacks seem pretty good, but a few fall too early or too late. Instead of simply quantizing the entire track so it snaps precisely to a rhythmic grid, you might try "percentage" quantizing.

A 50% global velocity increase yields greater dynamic range.

Depending on your sequencing software, you may have the option of quantizing only a certain percentage of the notes in a track, only a percentage of attacks that fall outside of a timing range (perhaps quantizing attacks that are more than 50 clock pulses early or late), or quantize with a percentage of strength. [*Ed. Note: While there are 24 MIDI CLOCKS per quarter note, most sequencers have higher internal resolution ranging from 192 to 480 pulses per measure.*] The concept behind using percentages is that some data is going to be altered and other data either will be left alone or modified to a different degree. The end result is that the track will sound more accurate, yet not quite perfect.

Even if math wasn't your strong suit in school, it's important to understand how percentages operate. Let's say that a track contains two delayed attacks: The first is 20 clock pulses late and the second is 36 "ticks" behind the beat. If these notes were quantized at a 40% strength, the first attack would occur at tick 12 (shifted eight ticks forward) and the second at tick 22 (moved 14 ticks forward). Both notes now are closer to the beat, but the human element is still maintained.

If your sequencer supports it, you can use these same percentage options when adjusting velocities. Increasing the velocity of a track by 50% means that an attack of 60 would be raised to the value of 90, while an attack of 80 would jump to the value of 120. In essence, this serves to open up and widen your dynamic contrasts.

Putting Some Kick Into Your Samples. Speaking of dynamic contrasts, it's a documented fact that the timbre and pitch of an acoustic drum change as the strokes get stronger. A bass drum, for example, has a higher pitch and a punchier attack when played with a lot of force. It's easy to duplicate this effect when using a sequencer along with a sampler. Here's how:

1) Take your bass drum track and find the velocity level for the strong accented notes. Let's say that a velocity of 110 is going to be the cut-off point (remember that 127 is the maximum).

2) Next, cut out all the notes with a velocity of 110 and above, and paste them into their own track.

3) Transpose the new track to a different note number and assign the same bass drum sample to that note.

4) Finally, use the controls on your sampler to slightly detune the loud sample by just a few cents (remember, this effect is subtle). You also may want to open up the filter slightly on these louder notes, to achieve a brighter attack.
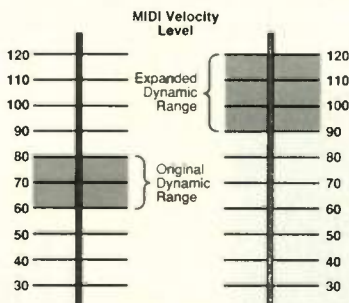
Double Your Options. Here's another little trick. Double-headed toms change their tone color as the bottom head resonates more and more with increased volume. To imitate this effect, create a duplicate of the tom track and assign this new track to a different sample. The sample played by the original track will serve as the sound of the upper head, while the sample played by the new track acts as the lower head. In the track assigned to the second sample, cut out any attacks below a certain velocity level so the more resonant sound only comes into play as the velocities increase.

The last step is to adjust the start time of the sample you're using for the bottom head. Since this sound functions as a resonating surface, you won't want to include the sound of the stick striking the head. Depending on your sampler, you may need to truncate the attack and/or alter the envelope to get the proper impression. You also may want to keep the second sample mixed low so that the effect is subtle.

Do you ever have trouble coming up with drum patterns that sound unique? Next time, we'll play a few games to give your creative juices a shot of adrenaline. •

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

Sequencers And Samplers Are Powerful Allies In Creating Realistic Drum Tracks.

GRAPHIC: RICK EBERLY

HANDS-ON PRODUCTION

Hip Hop Loops

BY MICHAEL MARANS

Pity the poor recording engineer. No longer simply required to know the ins and outs of the recording process, today's engineers now also must possess a thorough understanding of MIDI, multi-effects devices, multi-timbral sound modules, sampling, computer music software, SMPTE synchronization, and on and on. Working professionals know the frustration of not having enough "play time" in the studio to stay on top of new techniques and new technologies. Those who work primarily in personal studios often lack hands-on input from other recordists. So how can you stay on top of the latest techniques?

This column is here to help demystify high technology and teach you how to get the most out of today's creative tools. Welcome to "Hands-On Production."
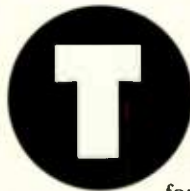
THEFT AS AN ART FORM? The current practice of sampling a piece of music from a CD and incorporating it into your song could be considered stealing. Practitioners of this form of petty pilfering prefer to call it *hip hop.* But we're here to tell you how to do it, not to make moral judgments. Let's just go on the record (no pun intended) and say that we don't condone this practice unless you obtain use rights from the original creators of your source sample. 'Nuff said. Now on to the techniques.

About Loops. The concept is quite simple: A small section of a rhythm track is sampled, then played back repeatedly throughout the course of a song. Although this technique is called "looping," the sample is not looped in the traditional sense of a looped sample. Rather, looping here refers to the fact that the sample is retriggered immediately after it has finished playing, which gives the impression that it's playing in a continuous loop.

Why not just loop the sound in the sampler? Despite your best intentions, it's nearly impossible to create a loop of the exact length required to repeat at the proper tempo over the length of a song without drifting. To do that, you need to trigger the sample as a sequencer event. And that's where the fun begins.

Michael Marans is a recording engineer, and an associate editor of Keyboard *magazine.*

Hip Hop Loop Chop

To calculate the loop length for known tempo at original pitch:
$[(60,000 \div bpm) \times \text{\# of beats in loop}] \times [\text{sample rate} \div 1,000] = $ loop length in samples.

Example:
$[(60,000 \div 119 \text{ bpm}) \times 4] \times [44,100 \div 1,000] = 88,941.175 \text{ samples.}$

$60,000 \div 119 = 504.20168$
$504.20168 \times 4 = 2016.8067$
$44,100 \div 1,000 = 44.1$
2016.8067×44.1
$= 88,941.175$

Preparing The Loop. Start with a sample that is one to four bars long. Since the rhythm within the loop most likely will contain a human element (a hi-hat part, for example), it's best to use the shortest sample possible. This will help you later, as it will minimize the amount of adjusting you'll have to do to make overdubbed parts fit in rhythmically with the loop's internal anomalies. (The longer the loop, the more anomalies there are likely to be.)

Now truncate the beginning of the sample so the transient of the first downbeat plays immediately when the note is triggered. Although you want this truncation point to be accurate, don't cut the beginning too short, as you may need to make further adjustments to get the loop to "groove." Truncate the end of the sample so no notes play after the end of the phrase. The box on this page shows how to determine the "ideal" length of a loop in relation to tempo. Once you're satisfied with the truncation points, enter the sample as a NOTE-ON event on the downbeat of a MIDI sequencer track. Quantize the event to fall on the track's first clock pulse, and enable the sequencer's track loop-in-play mode. The track should be the same length (in bars) as your sample.

Tempo & Timing. What if the sample is lifted from a song at 125 bpm and you want your song to chug along at 120 bpm? No problem. The tempo of the loop can be adjusted by transposing the sample up (to increase the tempo) or down (to decrease it). If you're just going for feel, it's a simple matter to adjust the transposition amount until the groove feels right. If, on the other hand, your loop must play at a specific tempo, trial and error (or, as the experts say, "experimentation") is the only way to set the proper amount of transposition.

Once you have the sample playing at the desired tempo, enter the true tempo into the sequencer (it helps to use a sequencer that offers tempo resolution in 100ths of a beat) and play the sample against the sequencer's click. Chances are, the loop will not play consistently in time with the click. Depending on the type and amount of transposition performed, the sample now will take a longer or shorter time to play, so you'll have to make adjustments to the truncation points to compensate for the time difference. If the groove is dragging, the sample is too long; if it's rushing, the sample is too short. You'll most likely have to work back and forth between adjusting the truncation points and the transposition amount.

The rules are simple: Transposition is used for tempo and feel; truncation is used for rhythmic integrity.

A New Art Form From The Streets Comes To The Studio— Here's How It's Done

THE STORY OF DAT TANGLED

TOO MUCH?

TOO GOOD?

TOO CHEAP?

TOO LATE?

BY CARY TENNIS

S OME WINTER STORMS IN SAN FRANCISCO blow down from Alaska, and some roar across the Pacific from Japan. The night I visited Different Fur Recording Studios, a big storm from Asia was blustering through the Mission district. I rushed from the wet, windy street into the studio. Co-owner Susan Skaggs, no relation to Boz, was waiting there to show me their DAT machines.

If the world were a simple and sensible place I wouldn't have gone out in such weather to see a Digital Audio Tape cassette recorder—a machine whose only flaw was that it did too much, too well, and too cheaply. DAT did not blow up or pollute the air or release

radiation. It did not smell bad or cause cancer, and it was too big for babies to swallow. All it did was make fabulous recordings on small, convenient tapes. Precisely because DAT performed so well, it was being withheld from American consumers. The first recording format to be developed entirely in Japan, DAT had been on the market in Asia and Europe for three years, but not here. In America, instead of retail sales, there were threats of lawsuits. There were congressional hearings involving expert witnesses, scholarly studies, and scientific surveys. Electronics and entertainment conglomerates, instead of cooperating to profit from a new technology, were at each other's throats.

The world is not a simple and sensible place. But it surely is interesting.

I had been up to my ears in DAT for weeks. I had heard the arguments, read the testimony, interpreted the survey data, gazed at the charts, studied the law, and been yelled at by all sides. But I had not yet seen the machine that

was at the center of the controversy. That's why I was standing, dripping wet, in the lobby of Different Fur.

I have always loved a good, dry, recording studio.

On the mixing console in Different Fur's main control room was a box of DAT tapes waiting to be labeled. In the rack was a Sony DTC-1000ES digital audio tape deck. The folks at Different Fur bought it in Japan about three years ago. "That's the first one that caught everybody's attention," said Skaggs.

The near-field monitors were Yamaha NS10Ms, which, like DAT, started out as a consumer product but ended up as standard studio fare. There also was the inevitable pair of Auratones. On a table sat a Sony DAT portable TCD-D10PRO. Then there was a Panasonic SV-3500 DAT deck. "That's the new one," said Skaggs.

We stood around and listened to some music on the DAT, switching from the near-field monitors to the main, custom monitors.

"More people now are mixing down to the DAT 2-track digital format, and less and less to 2-track analog," Skaggs said. "A year ago people still would mix to the analog 2-track and then make a DAT safety. It seems like almost overnight, people just fully accepted the DAT and feel confident in letting that be their master tape."

There you are. DAT is serious stuff. But you knew that. You may even have a DAT deck yourself, or have immediate plans to buy one. Twenty percent of masters arriving at pressing plants now are in DAT format. DAT is standard in the nation's professional studios. But in consumers' homes? That's another story.

As I walked back out into the rain, from the warmth and precision of the studio into the street, which was being washed clean of spilled beer and motorcycle oil and cigarette butts, I had a good sense of how DAT functions in the studio. But I still had no idea what was going to happen with DAT on the consumer market.

AS I LOOK BACK ON IT, I guess it was silly for me to call Sony and tell the receptionist, "I'd like to talk to someone who can tell me when consumer DAT machines are going to be released on the retail market." Given the secrecy that surrounded the subject until very recently, it was sort of like calling the CIA in Langley, Virginia, and asking for the names of agents in my area.

I finally reached someone in Sony's professional audio division who would talk to me. I asked him the big question: When?

"You won't find anybody in Sony who will give you that information," said David Kawakami. "They wouldn't even tell *me*," he said.

United States copyright laws are first for the good of the public—and second for the good of the individual creator.

THE DAT STORY REALLY BEGAN in 1790, when America adopted its first copyright laws. Copyright was a response to the printing press. Until it was possible to make multiple copies of a printed work quickly and cheaply, there was no need for a statutory right protecting "intellectual property."

Two hundred years later DAT represented the ultimate threat to music copyright. With DAT, theoretically, you could make an endless succession of perfect copies quickly, cheaply, and at home.

So the Recording Industry Association of America (RIAA), which is charged with looking after the rights and interests of the recording industry, came up with a standard policy: Until the problem of DAT's potential for copyright infringement was solved, the RIAA would sue anyone who imported DAT into the United States. In 1987, when Marantz, Casio, and other companies announced plans to introduce DAT to the U.S. market, the RIAA's lawsuit threats, combined with the record industry's reluctance to release music in a format whose future was uncertain, caused those companies to back down.

Finally, in June of 1989, by signing the "Athens Agreement," Japanese, European, and American entertainment and electronics interests cleared the way for consumer DAT. They all agreed to ask their governments to create laws that said all consumer DAT machines sold in their respective countries had to be equipped with the Serial Copy Management System (SCMS)—a system which, in essence, allows a consumer DAT machine to make a digital-to-digital copy of a CD, but doesn't allow further digital copying of that first DAT copy.

On February 22 of this year, Representative Henry Waxman (D-CA) sponsored the DAT bill in Congress.

The music publishing industry promptly hit the roof.

The Copyright Coalition—founded by the National Music Publishers Association (NMPA), the Songwriters Guild, and ASCAP—now vow to fight any DAT legislation that allows home taping of copyrighted material without also requiring a royalty tax on blank tape. The tax would benefit songwriters and publishers.

"To propose SCMS as a solution to the DAT problem is like giving the passengers of the Titanic life rafts that don't float," says NMPA president Ed Murphy. Morton Gould, president of ASCAP, says, "RIAA has been portraying this SCMS bill as a compromise supported by all affected parties. Nothing could be further from the truth."

Although the RIAA supports the concept of a royalty tax on blank tape, the Athens Agreement represents a working compromise with the Electronics Industries Association (EIA). And according to Cynthia Upson, director of communications for the EIA, the group opposes a royalty tax in any form. "I could never see us changing our position on this," she says.

Copy Codes For Copy Cats

A DAT DECK CAN MAKE a pristine copy of a CD—that's a fact. DAT decks must be prevented from making copies of CDs—that's what the record companies and publishing houses want. To date, there have been three copy-inhibit systems bandied about. Before we examine them, here's a bit of technical background:

All CD and DAT decks have traditional analog inputs and outputs; these I/Os add slight degrees of noise and distortion as they convert signals from analog to digital and back. Copy a copy of a copy of a CD, and the glorious sound of digital audio begins to fade. But some CD decks are equipped with digital outputs. Most DAT players also have digital outputs and some have digital inputs. (There are several different digital I/O formats; we'll explore them in the next issue of *EQ*.) CDs can be played back through the CD player's digital output and recorded through the DAT deck's digital input. This lets information *stay* digital as it's passed back and forth between machines. The result? Theoretically *perfect* digital-to-digital CD-to-DAT recordings—as many as you want. And that is what scares the record industry.

The simplest copy-inhibit scheme is a voluntary one, adopted by some manufacturers to import DAT machines into the States while minimizing the threat of reprisal from the record industry. Its premise is to prevent digital-to-digital CD-to-DAT copies, and it works like this: CDs are recorded and played back at a 44.1kHz digital sampling rate. The DAT format can record at 48kHz and 44.1kHz rates (and in some cases, at 32kHz). What Sony and others have done on some decks is allow playback at any of the sampling rates, but omit the capability of recording at the 44.1kHz rate—and this prevents a digital-to-digital CD-to-DAT copy. For recording studios, this system's biggest drawback is that DAT mixdowns are possible at 48kHz, but the music must take a quick detour down Analog Lane (through converters) to be mastered at the 44.1kHz CD rate. Fortunately for studios, a $200 modification adds the 44.1kHz recording capability to some of these DAT machines.

A "copy-code protection" scheme was the first system to go before the U.S. Congress. It called for the encoding of CDs with an "inaudible" frequency notch at 3,840Hz. DAT decks, in turn, would have been equipped with a chip that would detect the notch and turn off the recording process. In early 1988, the U.S. National Bureau of Standards concluded that the system sometimes didn't prevent copies, sometimes prevented recording when it wasn't supposed to, altered the sound of the music (surprise!), and could be bypassed easily.

And now along comes SCMS. The Serial Copy Management System, agreed to by manufacturers and record companies, is pending as U.S. law. Using a series of codes and "flags" in the inaudible subcode of CDs and pre-recorded DAT tapes, the system allows a single digital-to-digital DAT copy to be made of any copyrighted material (from a CD or pre-recorded DAT). You can make as many first-generation copies as you want from the master CD or DAT onto blank DAT cassettes. What you can't do is make a second-generation digital-to-digital copy of a copy; the DAT deck will recognize an "already recorded once" flag, and not record the material. Material without the "copyright flag"—including studio mixdown tapes and personal DAT tapes—can be copied digitally without restriction, using pro-style AES/EBU digital I/Os.

Under the proposals, all "consumer-oriented" DAT decks must be equipped with the SCMS circuit. "Professional" decks are not required to have built-in SCMS. Exactly what constitutes a consumer or pro model is somewhat vague. Tascam, for instance, recently debuted its DA-30 deck: It has *digital* RCA jacks that have SCMS; it also has *digital* AES/EBU XLR jacks without SCMS. Hmm. . . .

Some people see SCMS as a possible Trojan Horse: Once it becomes the law of the land, they feel, it is technically possible for record companies to release CDs encoded with the "already recorded once" flag. Forget about making any digital-to-digital copies, in that case. Says Jesse Jacobson of The DAT Store in Santa Monica, California: "Maybe there's already an unspoken agreement among the record manufacturers."

—Brent Hurtig

So the royalty tax battle remains to be fought. Perhaps after the question of SCMS legislation is decided, the RIAA will join the Copyright Coalition to battle the EIA on the matter of a royalty tax.

It's impossible to say exactly when DAT machines will hit the consumer market. But at the Consumer Electronics Show held in Las Vegas last January, Sony, Technics, JVC, and Denon all announced that they would have machines, with SCMS, available by early summer. What that will mean for pros is that as sales volume increases, DAT prices across the board should begin to fall. →

DESPITE ALL THESE THREATS AND AGREEMENTS, DAT machines have been perfectly legal in America, and around the world. This amazes some people. The *threats* of lawsuits are what have stymied the technology. Nevertheless, you *can* buy a DAT machine today, without having your face burn with shame. Fostex, Tascam, Sony, and other companies, even Casio, sell professional DAT machines. Many consumer-oriented DAT machines—those whose makers have faced lawsuits—are for sale in major U.S. cities through "grey market" dealers who import the decks, perfectly legally, from Asia.

THE COPYRIGHT QUESTION ITSELF HAS BEEN FOCUSED rather narrowly. A look at America's copyright laws suggests that home taping is in keeping with the larger intentions of copyright law. True, the laws do allow copyright owners to control distribution of their copyrighted material. But the main purpose of copyright law is to stimulate wide dissemination of creative products—*not* primarily to make sure that copyright owners get their buck.

Copyright is first for the public good—and second for the good (and well-earned property rights) of individual creators. So when it comes down to an apparent conflict between acts of dissemination (taping) and acts of remuneration (limiting taping and forcing people to buy originals), the courts often balance the larger purposes of the copyright statutes against the possible harm to copyright owners. They frequently rule in favor of broad dissemination, which is the underlying purpose of copyright.

What this really means is that the law has generally been on the side of private taping. The U.S. Congress passed a sound recording amendment to the basic copyright law in 1971 which stated, in part, "It is not the intention of the [resolution] to restrain home recording...[of] tapes or recordings of recorded performances where the home recording is for private use, and with no purpose of reproducing or otherwise capitalizing commercially on it." Subsequent revisions of the law in 1976 did not specifically address the home taping issue, and have been the subject of differing interpretations by RIAA and EIA. But the intent of the law is clear. We can't lock up and disable all the tape machines just to protect copyright holders.

BUT LET'S BE HONEST. RECORD COMPANIES and artists do lose some money to home taping. However, the best estimate is *three percent of potential revenue*. This comes from the U.S. Office of Technology's report, "Copyright And Home Copying: Technology Challenges The Law," perhaps the most thorough investigation of the subject to date. That report makes it clear that the much higher estimates of home taping losses touted by industry sources are unrealistic.

The world is changing faster than our written code of conduct.

The legal argument is that home tapers are actually stealing the intellectual property of the copyright owners.

Again, let's say yes, taping for any purpose other than personal use amounts to stealing. Remember the early days of personal computers, when most software was copy-protected? Users found that to be a pain. Companies found they had to weigh duplication losses against losses due to user dissatisfaction with copy protection. While companies struggled with copy protection, a curious thing developed among computer users: An ethic emerged in which users would self-regulate their software copying. Before making a copy for another person, they would weigh whether that person was a regular buyer of software, and how many copies they already had made of the copyrighted software. The communication by the software companies to users about the importance of copyright, combined with users' intuitive sense of right and wrong, made for a kind of organic, self-policed copy protection. It hasn't worked to the same degree as the methods used by BMI and ASCAP to protect songwriters' royalties, but it works nonetheless: All sorts of software companies are able to make big profits selling software programs that are not copy-protected. Could this work for music companies?

The psychic value of "the real thing" is another human factor that weighs against the likelihood of runaway home taping. Even if a tape sounds as good as the original, there still will be that desire to have the real thing. The perception that the copy is not as good as the original may persist even when the copy *sounds* as good as the original.

There are tangible and psychic rewards to buying the real thing. With an original copy of computer software comes a manual and update service. CDs come with a lyric sheet, photos, or at least a printed cover that authenticates the product. The fact that these are not as necessary to an audio product's usefulness as manuals and upgrades are to computer users is perhaps offset by the fact that audio recording in real time is significantly more of a hassle than just popping a floppy disk into a drive to grab an instant copy. By the way, DAT tapes, at up to ten to 15 bucks a pop, aren't exactly a cost-effective replacement for CDs, either.

Will SCMS really do anything to stop home taping? Some lobbyists conjure up visions of people stacking up DAT machines, all daisy-chained together, churning out endless copies of the latest albums. SCMS essentially says, "Ha! We'll put a stop to that nasty piracy by only allowing one generation." Have these people never heard of distribution amplifiers and parallel signal flow? SCMS is not going to stop serious lawbreakers, who would be much better referred to as "bootleggers" than "pirates." The revenue loss the recording industry is trying to fend off is from *serial copies*. The average person can make a copy for a friend. SCMS doesn't prevent that. Another friend wants a DAT copy of the same CD a week later. SCMS doesn't prevent that either. But if a friend of the *friend* wants a copy of the *copy*, that's where SCMS stops the buck..

SCMS also limits DAT-to-DAT, digital-to-digital copies, and may limit home taping in that respect. But in the legis-

lation's present form, it does nothing to prevent good-sounding, DAT-to-DAT *analog* copies from being made for several generations before their sound quality even approaches anything as "bad" as the media we've been listening to and recording on for years.

I N THEIR ZEAL TO PROTECT copyrights and profits, corporate policy makers have concentrated on the negative effects of home taping. They have been largely silent on the issue of how technology fosters the creative process that drives the music business.

Take rap. Rap is built around the ability to copy, to mimic, to manipulate previously recorded sounds; it began as a DJ skill, not an instrumental skill. It began because it was a cheap way for an underclass to appropriate the sounds of the dominant culture and turn them to expressive uses. It happened for the same reason that folk music happened: People without much money needed to express themselves. And no less a folk traditionalist than Pete Seeger has expressed excitement about the new "folk music" known as rap.

Now rap is a money-making phenomenon. But such money-making creativity does not spring full-blown from the corporate boardroom. It starts on the streets. And it starts by breaking rules.

Imagine if copyright holders had said to DJs who were mixing and scratching in urban dance clubs, "You can't use our music that way—you must play each song, beginning to end, as it was originally recorded. Unauthorized mixing and scratching violates our copyright!"

DAT will improve the quality of experimentation with musical forms. It is going to increase the quality of sampled sounds and put into the hands of fresh, creative spirits the tools to remake the world.

A S I WRITE, I AM LISTENING TO A CD of Depeche Mode, remembering a conversation last year with songwriter Martin Gore in Los Angeles. One of Gore's favorite samples is the sound made by striking the bell housing of a vacuum cleaner. It makes a rich, metallic sound, and samples of it appear throughout Depeche Mode's music. They use all kinds of samples, including pots and pans. We talked about how natural sounds are re-entering the sonic world through digital doors, so that one day someone cooking in the kitchen will hit a pan accidentally and think of a Depeche Mode song.

Technology changes things independently of our wishes. It is transformative. New technology does not just replace old technology; it redefines problems; it erodes the boundaries between functional niches. An acknowledgement of that seems to be missing from both sides of the DAT debate.

We have our own serial copy management system. It's called "conscience."

W HILE DAT HAS BEEN PREVENTED FOR THREE YEARS from gaining a foothold in the U.S. consumer market, researchers have been trying to develop a cheap re-recordable CD for the home, a format which some think could become more popular than DAT, because of its random-access capability and because it might be more durable than tape.

Tandy is developing THOR—Tandy High-intensity Optical Recording. The company announced in 1988 that it wanted to put inexpensive backwards-compatible (playable on existing machines) recordable/erasable CD technology on the market by mid-1990. Now it appears that 1995 is a more realistic goal. Sony is working on the Sony Magneto Optical Disk, which would use a combination of laser and magnetic recording technologies. In March of this year, Sonic Solutions introduced a $50,000 system that makes reference CDs [see "Update" section]. Still, the technical problems with recordable CD are legion. One of the chief hurdles is the fact that the quick access of CDs depends on their "tables of contents," the continual rewriting of which could get tricky.

The world is changing faster than our written codes of conduct. It will continue to do so. Our laws will not always be able to keep up. The ability to copy creative works perfectly has evolved from the printing press to the copying machine to computer disks to videotape to—heaven forbid—recordable CDs. Any legislation involving one isolated form of media will never solve the ongoing problem. Luckily, we humans are good at making judgment calls. That's how we survive. It's how we stay out of fist-fights on the highway; it's how we stay out of jail. It works for most of us.

We're capable of making the judgment call on DAT. We have our own serial copy management system. It's called conscience.

W HEN I WAS A KID, SOMETIMES WHEN we were about to sit down to eat there would be a family argument. Maybe it would be about how many miles it was to Washington, DC, from Nags Head, North Carolina. Before you knew it, there would be theories and data and testimony and threats of reprisals and counter-threats and it would seem like something much larger was at stake. For the people involved, I realized as I grew up, there usually *was* something larger at stake. But it didn't have anything to do with the distance from Washington to Nags Head, and it didn't have anything to do with dinner.

That's how I feel about this DAT business. The people who are doing the talking *do* have something larger at stake. Many of them may mean well. But what they have at stake may not have much to do with copyright and very well may not have much to do with music. Meanwhile the food's on the table and it's getting cold. We have waited long enough. Let's just sit down and eat. •

SELF-MARKETING

Marketing Without Money

BY JOHN SUGNET
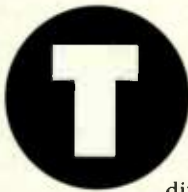
THE QUALITY OF ANY RECORDING has more to do with the art and craft of the people performing, producing, and engineering than it does with the equipment being used. Marketing isn't much different. If you have limited resources, then you just have to be more cunning about how you use them.

The most important marketing resource available to you is your own ingenuity. The old saying, "There is no such thing as a free lunch," definitely is true, but there are plenty of lunches that you don't have to pay for in cash. This is especially the case when you cross-promote with another party that has similar marketing aims. Here are a few approaches for you to consider.

Media Events/Cross Promotions. Some of the best marketing work I've done involved multiple parties with mutual interests. A grassroots-level cross-promotion might work like this:

You approach a local music retailer and offer to hold a seminar on a particular recording topic at their store. The retailer advertises your seminar in the store's newsletter and the local music rag, positioning you as an expert in recording. You get positive exposure to thousands of readers for free and a chance to form personal relationships with the people who attend the seminar. The retailer, in turn, gets to position his or her business as a resource for information as well as merchandise, and gets a serious group of potential buyers into the store to see a demo of the equipment being sold. The consumer gets access to information and expertise that is difficult to find. Everyone benefits!

You can apply this idea to a number of situations and target groups. Your event could be a benefit for a music-related cause or organization. It can be co-sponsored by an educational institution, a trade show, and so on. In some situations you can charge a small admission fee, which will pay for the time you spent preparing and making your presentation.

You do need to consider that with events of this type, two main things can go wrong: (1) The store owner does a poor job of promoting the event, and very few people attend; (2) You don't prepare carefully enough for the event and your presentation puts most of the people there to sleep.

Fortunately, neither of these sad situations has to occur. You can avoid the first problem by dealing with retailers who do a good job promoting their own business. Give them a good photograph, some well-written, grammatically correct copy that can be used verbatim, and plenty of background info they can use if they want to write their own promo copy.

The solution to the second problem is to prepare well for your seminar. If this is the first time you've spoken in public, you'll be more comfortable if you're well-rehearsed. Be prepared for things to go wrong. Pretend you're going on the road with a band; take two of everything, and scope out the venue carefully. Look for potential lighting, grounding, and audio problems. Remember, this is show biz; pump up the bass and give your show a strong hook.

PR & The Press. Once you've created an event, you're eligible for press coverage. Make sure to send a simple press release to all the magazines and newspapers that print events calendars. If you want greater exposure you'll have to work harder, but you can probably get it.

Day in and day out, journalists across the country sit in front of their computers and try to figure out how they're going to fill up the page. Help them! Being newsworthy isn't as difficult as you might think: You may not be able to create a story worthy of the *New York Times*, but you almost certainly can get into print somewhere valuable.

Adding to the example we used earlier (the recording seminar at the music store), you might put together a "Story Outline" on how inexpensive recording gear has revolutionized the demo recording business, or how an artist should choose a producer or engineer. Tie the story to your business and quote yourself. Include a Fact Sheet containing facts and sources that support your premise with the Story Outline. Put together a list of publications in which you'd like your story published, and shop them one at a time. Do not send this out to several editors or reporters at the same time. Call ahead and ask for the name of the person who should receive your press package, and follow up with a phone call asking whether they received your package and are interested in the story. Position yourself as a reliable source of information about the recording industry. Be personable but to the point. And above all, don't call press people just to chat. Don't get discouraged if your story doesn't gain immediate acceptance; you're still garnering experience and making contacts.

Like recording, marketing is creative problem-solving. The more you play with your marketing challenges, and examine them from all angles, the better and more elegant the solution.

Next issue: *getting graphic*. Do's and do not's for putting together ads. •

John Sugnet is a marketing specialist and consultant who has worked with many firms in the music and recording industry.

You Can Get Publicity And Press Coverage For Your Business— Without Paying For A Press Agent

didn't meet one of our requirements—that of low price (the monitors we tested all cost under $1,000). For looped program material, we used: the Shep Pettibone remix of the track "Hard Day" performed by George Michael (on the Faith CD, Columbia, DDD) for dance/pop music; the track "In The Still Of The Night" from Whitesnake's self-titled CD (Geffen, ADD) for rock music; and, for solo acoustic piano, the cut "By The Fireside" performed by William A. Mathieu on the CD A Winter's Solstice II (Windham Hill, DDD).

I've read Larry Blake's column "The Challenges Of Post-Production" in the premier issue of EQ (which, by the way, is great!). I am a 19-year-old college student in Seattle, Washington. The art of film and video sound production interests me greatly, but my school does not offer courses in this. Can you recommend any schools, courses to take, academic majors, and job opportunities available in this field?

I have no musical talent to speak of, but is that a prerequisite for this field, or any other recording job? Thank you.

KEN SMITH
Seattle, Washington

Because of limited space, we can't talk about the many job opportunities available in video and film production, but there are plenty. A trip to your local library for some research might be in order (look in the card catalog under "video" and "film"). You might talk with your college's career counselor, who should have information about what's available in your area. Many large universities and colleges offer course studies in film and video, with degrees in video production, film production, broadcasting, and mass communications. Get catalogs from Washington's state universities—there might be something there. Also, the Mix Bookshelf sells educational books and videos that mainly concern recording, but they have material on video and film, too. Call (800) 233-9604 for a free catalog.

You won't get too far in the music recording business without some knowledge of music and musical theory, but that's not to say you must be a violin virtuoso. And there are many people in video and film who don't know a thing about music. However, the audio and video disciplines are merging, and will continue to merge. If you're looking to get into one of those areas, it's best to familiarize yourself with both. Keep reading EQ for more insights into the future of audio and video. And good luck!•

IN THE POCKET

What's In A Name?

BY STEVE HARLAN

THE APRIL 15TH, 1990 TAX filing deadline is quietly sinking into the sunset—dragging along with it, perhaps, more of your hard-earned studio and music biz profits than you care to think about. Taxes might be a sore subject right about now, but I promise not to throw salt in the wound. Instead, I've got some tax ideas that just might save you some real dough!

Before I get to the point, I must point out (as always) that you should consult with a tax professional before taking any action on what you read here.

Let's give some thought to the different legal and tax choices regarding the form in which you operate your studio or music business. These can make a big difference in your tax and legal liabilities.

Basically, there are four choices of form for conducting your music business: as a *sole proprietorship*, a *general partnership*, a *corporation*, or an *S or tax-option* corporation. We'll discuss the first two in this issue.

Sole Proprietorship. A sole proprietorship (single owner) is the least expensive form of doing business, from the standpoint of legal and accounting fees, and general administration.

The advantage of a sole proprietorship, as far as taxes go, is simplicity. If you're in the studio or music biz as a sole proprietor, then you probably already know that all you do is attach a schedule "C" to your tax return to report your business income or loss. As for legal requirements, as a sole proprietor, you're the boss. There are few government regulations to concern you (beyond the usual things such as business licenses and building codes), and no corporate boards of directors or partners to answer to.

The advantages of the "sole prop" form of doing business, however, stop with simplicity and lower professional fees. Depending on your situation, there could be a number of disadvantages to a sole proprietorship. For example, the business must report taxes on the same tax year as the owner; in other words, December 31. As a separate tax entity like a partnership or corporation, your studio business may be able to adopt its own fiscal year. Frequently, this can open up all kinds of tax planning possibilities.

Although a "sole prop" can have a retirement plan (Keogh or SEP plan, for instance), the retirement deduction saves income taxes only, not self-employment (social security) taxes, unlike a corporate plan. The social security taxes for 1990 are currently 15.3 percent.

Finally, there are numerous tax-favored fringe benefits that are not available for sole props. Among these are tax-qualified medical plans, tax-free meals and lodging for travel, and so on.

The Partnership Form Of Doing Business. Now, here's something that's bound to interest the typical group of individuals conducting a studio, band, or other creative endeavor. If you're not the sole owner of your business, you generally do not qualify to operate as a sole proprietorship for tax purposes.

Therefore, unless incorporated, the business generally would be considered a partnership by the IRS, even if no written agreement of partnership existed. The IRS can impose some hefty penalties for failure to file a partnership return (Form 1065) when one should be filed.

Concerning limited liability, a partnership is like a sole proprietorship. In fact, it's even worse. Say your partner decided, without your knowledge, that your studio should acquire a hippo from the zoo to get a few new digitally sampled grunts and other obscene sounds. You could be held personally liable for the cost of the hippo plus any damages incurred, such as general mayhem in transporting the hippo to your studio. If the hippo escapes and runs rampant through all the businesses on your street, you could have quite a bill on your hands.

A partnership is what's known as a "flow-through" tax entity. Basically, this means that partnership income or loss (whether or not it's actually paid out to the partners) "flows through" proportionately to each partner and is taxed to that partner as individual income. The partnership itself, unlike a corporation, pays or withholds no tax.

(Tax Hint #1: *This "flow-through" feature can be very advantageous in business start-up situations where initial operating period losses will occur. The losses are deductible by the individual partners.*)

There is an important disadvantage to operating a studio business as a partnership. Many favorable tax deductions, as you may be aware, are limited in dollar amount. These limitations may be imposed at the partnership level rather than at the individual level. For example, the partners would have to share the special first year $10,000 expensing election on studio equipment.

(Tax Hint #2: *Individual ownership of equipment by the partners, with the equipment leased to the partnership, can provide significant tax benefits.*)

Next time we'll look at the pros and cons of operating a recording studio or music-related business as an "S" corporation, a "C" corporation, and as a limited partnership. •

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

Make The Smart Choice About Your Studio Business' Organization— And Reap Big Dividends

ARRANGING

FOR THE STUDIO

A Session Primer
For Producers And
Engineers
By Ernie Mansfield

SHMRG (pronounced "shmirg") isn't some foodstuff from Jupiter. It's an acronym for "Sound, Harmony, Melody, Rhythm, and Growth" that music theorists use when they discuss music. *Sound* describes the type of musical colors used in the composition. *Harmony* and *melody* are terms used to describe the musical shape in the vertical and horizontal planes. *Rhythm* can be defined as sound's relationship to time, and *growth* is an explanation of how the piece of music unfolds over time.

So what does SHMRG have to do with recording? Everything! A music arranger is aware of how each of these five categories interacts and relates—and engineers and producers should be, too. Each aspect needs to be tied into the musical whole before that vague, ethereal "sound in your head" becomes the music that gets recorded, produced, and packaged in tangible form. Whether you work with other "real" musicians—or work strictly with synthesizers, samplers, and drum machines—arranging is a skill that will improve your craft work. →

15 Arrangements That Broke The Rules

Charlie Parker, *The Savoy Sessions*, 1945. The definitive bebop group of Parker, Dizzy Gillespie, Miles Davis, pianist Argonne Thornton, bassist Curley Russel, and drummer Max Roach. Parker's chromaticism radicalized jazz; his syncopation made the bar lines jump right off the music stand.

Miles Davis, *Sketches Of Spain*, 1959. Arranger Gil Evans mixes unusual horn combinations like no one else (all his albums with Davis are wonderful). *Sketches* broke more rules by blending flamenco and jazz.

Oliver Nelson, *Afro-American Sketches*, 1961. Before becoming a Hollywood arranger, Oliver Nelson was a great jazzman. *Afro* features ethnic motifs arranged for a big band combined with strings, French horn, and tuba.

Jimi Hendrix, *Are You Experienced*, 1966. Hendrix explored the use of amplifiers and tape machines as instruments, and volume as an arranging tool. *Electric Ladyland* also broke rules with its use of multitracked guitar parts. Guitar simply hasn't been the same since.

The Beach Boys, *Pet Sounds*, 1966. Brian Wilson's surf-pop chorales would put him on any

A Little History

IN WESTERN CLASSICAL MUSIC, composers like Mozart, Beethoven, and Brahms wrote out their pieces note-for-note for specific groups of instrumentalists and/or vocalists. The composer also was the arranger and exercised total control over the content and organization of the piece.

In American popular music of the '20s, '30s, and '40s, tunes were written by songwriters, then scored by staff arrangers and performed by staff musicians. This was the legendary "Tin Pan Alley" method of working, where the composer and arranger were two different people.

In the '50s, when folk and country-western music was first recorded, record companies sought out singing groups rather than the traditional, individual composer, arranger, and performer. These groups worked out their material by ear, coming up with ideas and trying them out as a group, until they found the desired arrangement. Often, record companies hired professional staff arrangers to write background orchestral or vocal parts to "sweeten" the sound. This is still done today. But in a typical recording session, today's groups record their own self-composed, self-arranged music.

The real truth of the relationship between arranging and recording is that *everything you record has an arrangement*—even if the arranging is done unconsciously by those who "don't know how" to arrange.

The art of "conscious" arranging can bring order and organization into a recording session. By working out an arrangement in advance, one can schedule studio sessions more efficiently, by knowing how much time is needed to record, how many and what type of musicians are needed, how many tracks will be used, and so on. As a producer or engineer, the greater your arranging skills, the more the musicians can devote their creative energy to the performance. And even if you're a solo musician/engineer, a basic grasp of arranging skills can add realism, dynamics, and professionalism to your work.

Given endless amounts of time and money, just about anyone can make a decent recording. But a good arranger can do it quickly and efficiently.

Fortunately, you don't need a complete course in music theory to learn some concepts you can use right away.

Sound = Musical Colors

THE MORE YOU KNOW ABOUT the characteristics of an instrument, the better you will be able to arrange for it. One thing an ar-"range"-er needs to know is the ranges of the instruments being used. Asking an instrumentalist to play a note that is too high or too low for the instrument creates major hassles. In addition, all instruments have idiosyncrasies that must be addressed.

For example, certain patterns of notes may be easy for one instrument—and represent a real finger-twister for another. String players prefer keys that use sharps because of the nature of the fingerboard: The open strings of most stringed instruments include *G*, *D*, *A*, *B*, and *E*, and these all happen to be sharp keys. On the other hand, brass players prefer keys with flats in the key signature. Horns (brass) and single-reed wind instruments (clarinets and saxes) are transposing instruments, meaning that the note that is read and fingered as a *C* is not actually a concert *C*, but the pitch of the instrument. So when a B♭ trumpet player reads and fingers a *C*, the note you hear actually is a concert B♭. This means the key of *E*, a great key for strings, would be horrible for B♭ brass players. (It would be written in the key of *F#*—six sharps—aaggh!)

In addition to the transposition matter, there are timbral considerations. For example, the clarinet sounds rich and full in its lowest register, bright and piercing in its upper register. Whether or not you end up scoring for these instruments, it is helpful to know about the frequency ranges of the various instruments as an aid when recording. [*Ed. Note: Check out our "Studio Reference*

Ernie Mansfield is a freelance arranger and recording artist. His latest album is entitled Color Drops *(on Catero Records). His business, Music Graphics, specializes in preparing music for publication.*

ILLUSTRATIONS: RICK EBERLY

Tips For Arranging

1) Listen to music in the style you are trying to arrange. Transcribe an example for yourself: First, write out the basic form; then, identify the instruments, chords, and melody line; finally, write out the actual notes being played.

2) Plan arrangements well in advance of the studio session. Make rough demos (cheap synths have their place!) in your personal studio.

3) When layering or overdubbing instruments, listen to the playback after each addition. You may want to add fewer instruments than originally planned. A "thinner" score may produce cleaner, clearer instrumental lines, while a "thicker" score may be muddy and indistinct.

4) At the session, solicit the opinions of musicians and others responsible for the production.

5) If an idea is just not working in the session, don't force it. Go on to something else. You'll be surprised how fast a new idea will come to take its place.

6) If the musicians require clear, written parts and your music writing skills aren't up to the task, hire someone to do this. Conversely, if the musicians prefer to play by ear, don't bother with complex written parts: Provide a simple demo tape.

—Ernie Mansfield

Series" on page 76 for a chart of instrument ranges.]

Vocal range is another major consideration. In typical pop music, the singer's range is about an octave plus one step (or a ninth), considerably more limited than most instruments. Classically trained vocalists often have much broader ranges. Since most instrumental tracks are recorded before the vocal—unless every part is MIDI-sequenced (which makes for easy transposition)—it would be a disaster to record a full-blown arrangement and find that it had to be re-done in a different key for the singer! While it's easy to make a general range chart for voices, every singer is unique, so be sure to rehearse with the singer beforehand to determine the best key.

Listed in the credits of most major motion pictures is someone called the "orchestrator." This person takes the composer's music and literally scores it for the instruments. Although the music you're working on may include instruments not found in the traditional orchestra, you'll find it helpful to analyze the orchestra in terms of instrument families and how they are balanced.

The traditional orchestra is divided into these families: (1) Bowed strings: violins, violas, cellos, and basses; (2) Plucked strings: harp, guitar, mandolin, lute; (3) Woodwinds: flute, oboe, clarinet, bassoon, saxophones; (4) Brass: trumpet, trombone, French horn, tuba; and (5) Percussion: drums, cymbals, timpani, gongs, mallet instruments (xylophone, marimba, vibraphone, glockenspiel), and miscellaneous hand percussion.

At mixdown time—whether the parts are "real" or synthesized—a common challenge for the engineer is to create a realistic placement for instruments in the mix. The traditional orchestral seating arrangement (see Fig. 1) offers some insight into the number and placement of the instruments, which together achieve an acoustic balance in a live setting. Notice how the strings greatly outnumber the woodwinds and brass, and how the percussion and bass instruments are positioned behind everyone else. [Ed. Note: There is some variation among orchestras. Sometimes the harp is placed in the back near the piano; some orchestras, such as the Boston Symphony, juxtapose the violas and celli to concentrate more low-frequency tones in the middle of the ensemble.]

Orchestral writing favors the strings, which play predominantly throughout most of the work, with parts performed by many players in unison. Doubling an instrument does not just increase the loudness; it causes a thickening of the sound, and a masking of intonation and tonal differences—the same principle as "chorusing" instruments in the studio by electronic means. The basses remain in the rear of the orchestra, as their sound is perceived to "spread through" the ensemble. Brass and percussion also are located in the rear, since their sounds punctuate those of the strings and woodwinds. Woodwinds are closest to center stage because their sounds are more delicate; they're located near the strings, since they often are used to double with the string

arranger's "faves" list. Pet Sounds—featuring found objects like bicycle bells along with such instrumental combinations as banjo and kazoo, and vibraphone and orchestra (and the first major use of synthesizer in pop music, on "Good Vibrations")—is his instrumental masterpiece.

The Beatles, Sgt. Pepper's Lonely Hearts Club Band, 1967. The ultimate rock band made 4-track magic, and turned the tape recorder into a pop arranging tool. John Lennon once asked, "If George Martin's such a genius, why don't people buy his albums?" But hey, at least with this group, Martin was a genius.

Walter (Wendy) Carlos, Switched-On Bach, 1968. The first popular album performed completely on a (monophonic!) synthesizer, Switched-On Bach made the word "Moog" a household word overnight. While Carlos' rather straightforward timbral manipulations worked well with Bach's contrapuntal style, the public assumed this is what a synthesizer sounded like until other rule-breakers followed. Another contrast of traditional composition and synth was Isao Tomita's marriage of abstract timbres with Debussy's impressionist compositions on Snowflakes Are Dancing. Along the way, Keith Emerson first used synth as a rock solo instrument on "Lucky Man" (ELP), with Jan Hammer adding the pitch wheel on Mahavishnu Orchestra's Birds Of Fire, and Pete Townshend adding rhythmic arpeggiation

Fig. 1.
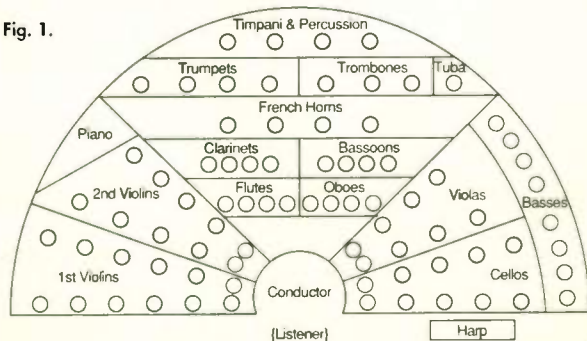
on the Who's "Baba O'Reilly" (*Who's Next*) .

Broadway Cast, *Hair*, 1968. Rock's first foray (musically and culturally) into operatic theater form. The songs (including stellar hits like "Aquarius," "Good Morning Starshine," and of course, "Hair"), sung onstage by actors and actresses, formed part of a larger story. This branch of the musical tree was later to bear such fruit as *Tommy*, *The Rocky Horror Picture Show*, *Jesus Christ Superstar*, and *Little Shop Of Horrors*.

Yes, *The Yes Album,* 1971. This British group's first international offering was seminal in the art-rock genre shared by groups such as ELP, Moody Blues, Pink Floyd, Genesis, and Gentle Giant. Classical background brought concepts like song movements ("Starship Trooper"), deviation from 4/4 time ("Yours Is No Disgrace"), and contrapuntal technique ("Perpetual Change") to the rock arena. Yes was to follow with other definitive arranging landmarks such as Simon & Garfunkel's "America" (B-side of the "Roundabout" single) and one of the first album-side-long songs, "Close To The Edge" (*Close To The Edge*).

Weather Report, *Weather Report,* 1971. Building on Miles Davis' *In A Silent Way* (its title tune penned by Joe Zawinul), the group explored textural themes while reducing the soloist's role in jazz. Along with Larry Coryell and Miles

passages.

Within this physical structure, the orchestrator tends to score instrumental "families." The string section's music makes sense harmonically when played alone, and when the brass make an entrance, they also harmonize among themselves. A sheet of orchestral staff paper illustrates this style of orchestration (see Fig. 2), because it lists the instruments by family groups.

Harmony = Musical Shape

ALMOST ALL WESTERN MUSIC uses *harmony*. When two or more voices (instrumental or vocal) join together, unless they sound the exact same notes, they create harmony. Two voices form intervals; three or four voices form chords. While two-voice intervals may not necessarily give the strongest sense of tonality, three or more voices inevitably produce a sense of key and tonality. For example, the simplest modern chords consist of at least three notes: root, third, and fifth (e.g., a C Major chord = $C\,E\,G$; see Fig. 3). The most common form of harmony is four-part. In this case, the root (C) may be doubled, producing the chord ($C\,E\,G\,C$). If the interval of the seventh is added, the chord may be spelled ($C\,E\,G\,B\flat$).

Four-part harmony manifests itself in a lot of popular music we hear today. Vocal music (soprano, alto, tenor, bass) and the string quartet (violin 1, violin 2, viola, cello; see Fig. 4) are obvious examples. Other interesting combinations include: (a) guitar, keyboards, bass, drums; (b) lead guitar, rhythm guitar, bass, drums; (c) solo piano; and (d) fiddle, mandolin, guitar, bass (see Fig. 5). A skilled arranger possesses extensive knowledge of chords, their substitutions (which chords can replace others), voicings (how closely or widely spaced are the notes in the chord), and so on.

When two or more chording instruments are used, such as piano and guitar, it is *essential* that they keep out of each other's way. It's so easy to create confusion and "muddiness" in an arrangement by having these instruments play similar voicings of the same chord. Thanks to the nature of piano and guitar, there are many inherently different ways to voice chords—use them! Even two people playing the same instrument have conflicting ideas on chord voicing—which is why a typical rock band has lead *and* rhythm guitarists.

Melody = More Musical Shape

WE'RE ALL FAMILIAR with the concept of melody. A strong melody can be the single most important factor in a song's success. In addition to the main melody, many compositions contain *counter-melodies*. Counter-melodies (or counter-

Fig. 2.
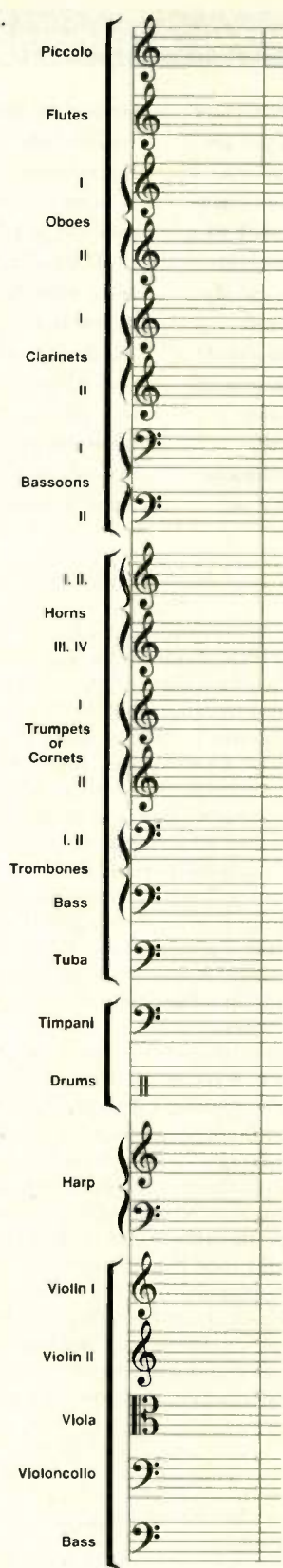
point) occur when two or more voices appear to have their own independence. The simplest form of "contrapuntal" music is a *canon*, such as "Row, Row, Row Your Boat" (see Fig. 6). This tune has a single melody,

Fig. 3.

C major chord

Fig. 4.

c root (1st) = Soprano = Violin 1
g fifth = Alto = Violin 2
e third = Tenor = Viola

c root (1st) = Bass = Cello

Fig. 5a.

Guitar
Piano
Bass
(Unpitched) Drums

Fig. 5b.

Lead Guitar
Rhythm Guitar
Bass
(Unpitched) Drums

Fig. 5c.

Fig. 5d.

Fiddle
Mandolin
Guitar
Bass

but when the next voice enters singing the same part, the counterpoint effect occurs.

On a more complex level, try listening to the fugues (for keyboard or solo violin) of J.S. Bach and follow each independent line. Bach was a master at weaving a tapestry of imitative, independent lines to form a piece that is still harmonic in nature.

Although contrapuntal music was in vogue during the Baroque period, it is quite common today. Listen to a Dixieland jazz ensemble and notice how each instrument is both a "soloist" playing its own melody, and an accompanist providing support for the other instruments. Another use of counterpoint is a counter-melody in the background of a song, with the primary melody stated in the foreground. Most pop ballads use this effect.

Rhythm • Sound's Relationship To Time

RHYTHM CAN BE APPROACHED on two different levels. On the small scale, the amount of rhythmic activity can make a song sound hectic or relaxed. How the individual parts interact rhythmically can produce a feeling of unity *or* conflict.
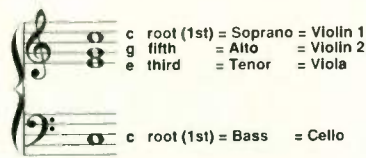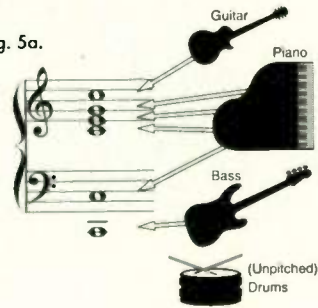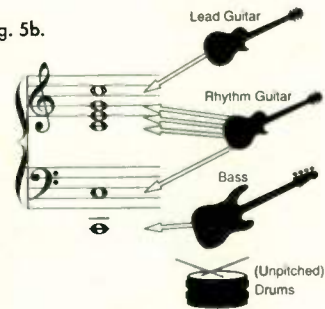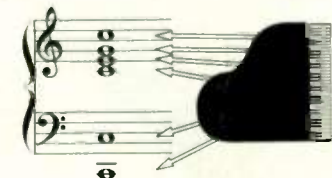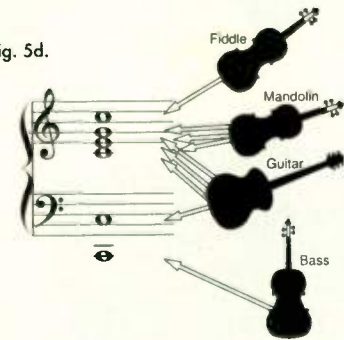
On the larger scale, each section of a work takes up time. The relationships between lengths of the different sections also contribute to the impression of the song. Who could forget the na-na-na chorus at the end of "Hey Jude," or Stevie Wonder's repetitive solo closing "Isn't She Lovely"?

Rhythm also can *define* an entire arrangement: Listen to "Hey, Bo Diddley" by Bo himself, or "Sing, Sing, Sing" by Benny Goodman with Gene Krupa on drums.

Growth • Music Unfolding Over Time

MUSICAL "FORM" IS ESSENTIAL-LY the structure of a piece of music, and only a handful of forms are used in popular songs. The most common is called *song form*: A-B-A, or Chorus-Verse-Chorus (see Fig. 7)—a song with several alternating verses and choruses. A variation is A-B-A-C-A, or Chorus-Verse-Chorus-Bridge-Chorus. If you add the concepts of "introduction" and "coda" (an ending), it is easy to understand the concept of form in most of today's pop and rock songs.

A successful musical composition relies on a balance of *unity* and *contrast*. Too much unity makes a work boring and predictable. Too much contrast creates chaos and confusion.

To be understood and remembered by the public, popular music relies as much on repetition as it does a memorable hook. Knowing what and how much to repeat is an important part of successful arranging, for musical as well as pragmatic reasons. For example, a standard pop tune is about three minutes long. Creating a longer tune might be tempting, but probably won't get commercial radio play. A jazz recording, on the other hand, almost always includes many repetitions of the A-B-A form so each instrumentalist can take solos over the top.

Davis alumni such as John McLaughlin, Chick Corea, and Herbie Hancock, Weather Report set the direction for electric jazz and fusion.

Mike Oldfield, *Tubular Bells*, 1974. The first new-age record. Trance-like repetitive passages, combined with traditional instrumentation, held the key to a musical form that would take more than a decade to grow from an anomaly to a popular, alternative musical form.

Kraftwerk, *Autobahn*, 1974. The first really mechanized commercial sound from Germany spawned two different mainstream offspring: disco (typified by Georgio Moroder's work on Donna Summer's "Love To Love You Baby") and techno-poppers such as Gary Numan, Soft Cell, and Devo. These two styles were to merge again as the dance music of the late '80s.

Jaco Pastorius, *Jaco Pastorius*, 1976. Jaco redefined the role of the bass in arranging. Highlights include a cooking duet with congas on Charlie Parker's "Donna Lee," and the incredible "Portrait Of Tracy," an original solo piece in which Jaco magically conjured bass, melody, and chords (via harmonics, no less) from a lone bass in a single pass of tape.

Bobby McFerrin, *Bobby McFerrin*, 1982. The oldest instrument (the human voice) and one of the newest (the tape recorder) were combined by one man to create all the parts of a traditional arrange-

Manipulation of the musical form over time, known as "linear arranging," represents an important decision. By establishing a piece's form, you also determine the length and dynamic action of the performance. Will it start soft and build to an exciting ending? Will it start moderately, build to an exciting bridge, then soften to a repeat of the previous dynamic? Listen to pop standards, and you'll hear formal structures that create successful arrangements. Some structures may include an instrumental break used in place of a verse, or a modulation to a higher key for a final chorus.

The Arranger & Technology

LANDSCAPE PAINTERS CAN SCRAPE one color off the canvas and replace it with another. Sculptors can remove clay if their statue doesn't have the proper form or structure. But composers and arrangers once had to hire additional people in order to hear their compositions. If it wasn't quite what they wanted, it was back to pencil and paper, requiring evermore time and money.

Now, thanks to synths and samplers, it's possible to experiment on many different levels until you're completely satisfied. How would it sound with an English horn doubling the vocal line, a harpsichord replacing the acoustic piano, a set of tuned bottles added to the chorus? How is the overall musical impression altered if the introduction is shorter, the chorus twice as long, or the bridge slowed down? With MIDI technology, the answers to these questions come easily.

Yet this technology is not without its pitfalls. A keyboardist firing violin sounds is going to sound like a keyboard player, unless his or her playing technique is modified to emulate a violinist. This is another reason it is vital to have knowledge of orchestral instruments and their stylistic performance techniques.

The sounds and colors of the new technology also can join the arrangement. One popular

Fig. 7a.

Fig. 7b.

Fig. 7c.

{* Hook line from chorus, repeats and fades}
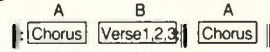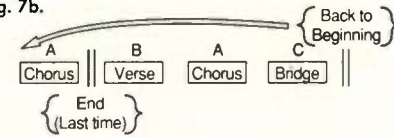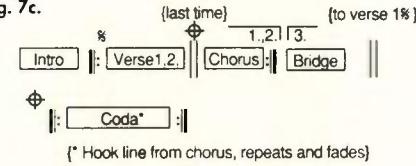
technique involves doubling real acoustic instruments with synths and samplers to gain a thicker, more textural sound. [*Ed. Note: This technique is used by producer Chip Davis; see page 18.*] Another technique involves layering two or more synthesizers to achieve a sound with richer harmonic content. As electronic instruments continue to break sonic rules, the progressive arranger must borrow and modify traditional SHMRG elements that often are taken for granted.

Recording techniques also can directly influence arrangements. For example, using reverberation effects, such as small or large halls, affects the performance. At times, a unique outboard effect represents a major contribution to the arrangement; listen to the classic gated drums in Phil Collins' "In The Air Tonight.".

A good arranger will determine what tracks to record first, and what should be overdubbed later (perhaps even after the vocals). Sometimes a "live feel" between players or singers is necessary, and the advantages of overdubbing are outweighed by the benefits of live musical interaction.

As with any art, arranging can be perfected with practice. The arranger's goals are to maximize efficient use of studio time, improve the overall quality of the music and recording, and make decisions early in the session to avoid the "fix it in the mix" syndrome. Given endless amounts of time and money, just about anyone can make a decent recording. A good arranger can do it quickly and efficiently. •

Fig. 6.
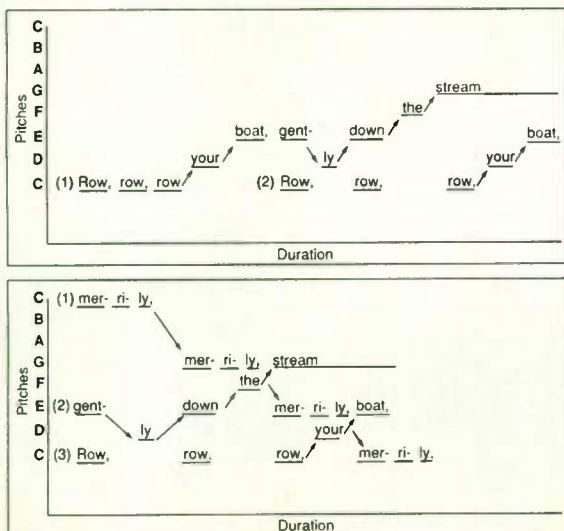
ment. Vocal kudos also go to Manhattan Transfer and Jon Hendricks for jazz vocal-group arranging, and to Queen for vocal pyrotechnics on *A Night At The Opera* ("*Bohemian Rhapsody*").

Kronos Quartet, *The Music Of Thelonious Monk*, 1984. Spare and angular bebop melodies, arranged for string quartet and quintet. Tom Darter's arrangements, and the purity of the string quartet form, uncovered the composer's timeless genius.

And Let Us Not Forget:
Dave Brubeck Quartet, *Time Out* (use of non-standard time signatures in a jazz quartet); Rob Wasserman, *Duets* (arrangements for voice and bass); The Ramones, *The Ramones* (an exercise in *anti-arranging* that inspired punk, and hardcore); Living Colour, *Living Colour* (combines funk, thrash, metal, and chops); Peter Gabriel, *Peter Gabriel* (gated drums, no cymbals); Kate Bush, *The Dreaming* (seamless intertwining of textures); Prince, *Controversy* (use of rhythmic *everything* with *no* reverb); Michael Jackson, *Thriller* (Quincy brought in Van Halen and suddenly every R&B tune had pyrotechnic guitar); Jean-Pierre Rampal & Claude Bolling, *Suite For Flute & Jazz Piano* (two genres meet head on); Brand X, *Brand X,* (rockers playing jazz); Sting, *Dream Of The Blue Turtle* (jazzers playing rock); Frank Zappa, everything from *Hot Rats* on (what else can you say about Zappa?).

—Phil Hood, Jeff Burger,
& the EQ staff
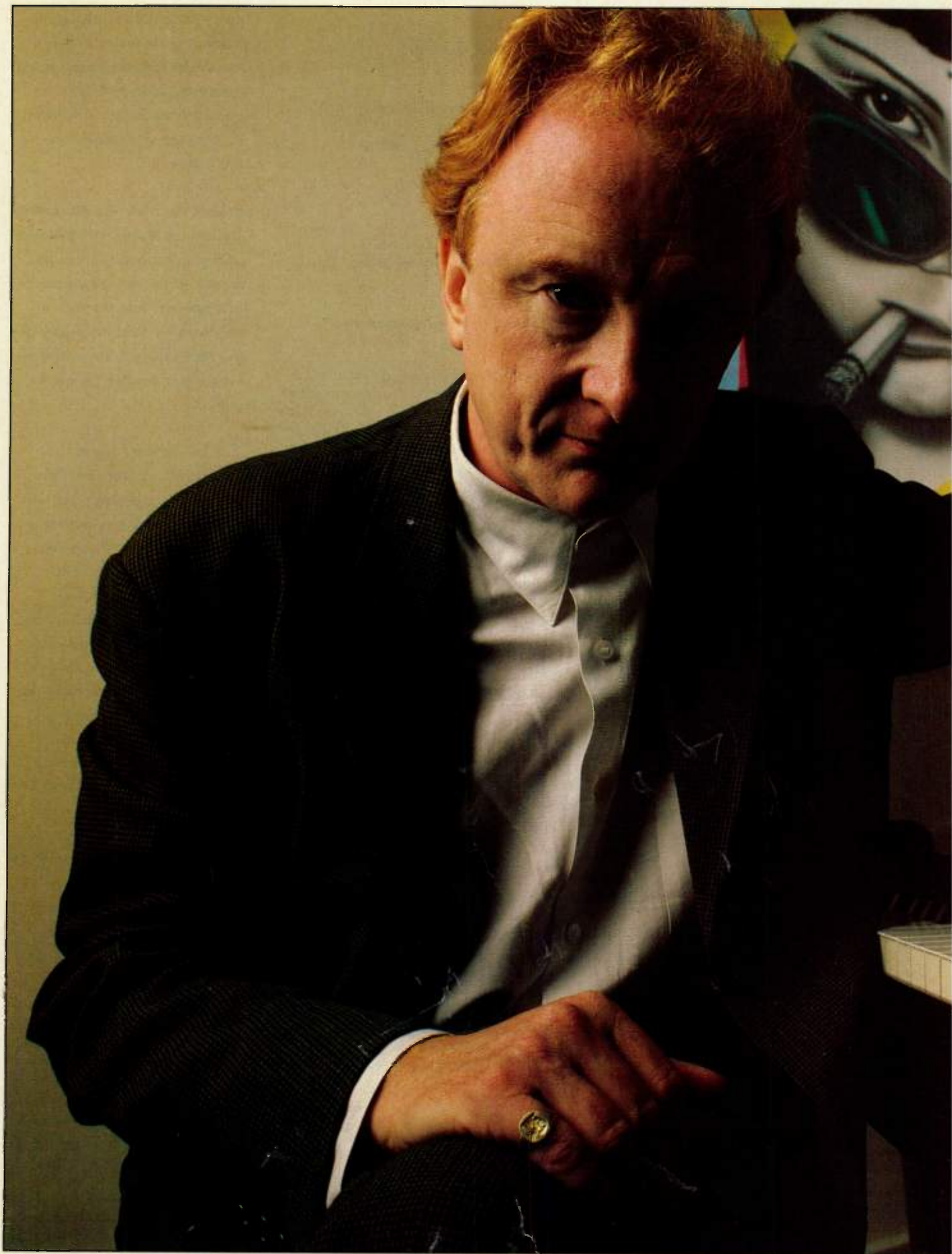
Peter Asher

Producer Of The Year

Linda Ronstadt, James Taylor, And 10,000 Maniacs Can't Be Wrong. By Amy Ziffer. Waiting in the reception area of Peter Asher's airy West Hollywood office, I had a moment to reflect on the man's accomplishments: the first head of A&R for the Beatles' Apple Records; 25 gold and 15 platinum albums (including five Grammy award-winners) on the wall by the entrance, almost every title a classic; two Grammys for Producer of the Year (this year, and in 1978). And all that only represents half his work. Rows of cassettes on a shelf bear the names he manages: James Taylor, Joni Mitchell, Randy Newman, Carole King, Little Feat, Linda Ronstadt, and new artists who have yet to make

James Taylor, *Sweet Baby James,* 1969

"*Sweet Baby James* was a record that was made fast and cheap. We knew we had to make it reasonably cheap, but I felt guilty about how cheap it actually was. Carole [King] and Russell [Kunkel] and the bass player [Lee Sklar] would come over to my house in the afternoon with James and we'd rehearse two or three songs in the living room, then go into the studio that night and cut them. It was all too easy. The album cost $8,000 and our budget from Warner Bros. was $40,000, which now, of course, would be a drop in the ocean. But at the time I felt guilty that we'd only spent eight out of the 40."

Linda Ronstadt, *Heart Like A Wheel,* 1974

"*Heart Like A Wheel* was a record we attacked one song at a time, and really tried to make it the way we wanted it. It took a while, but it was worth it. We cast each track with musicians, and we spent a lot of time on some overdubs, like on "You're No Good." Andrew [Gold] and I stayed up way into the morning doing all the guitar overdubs. Linda came in the following night—she'd gone to bed at a normal time—and hated them. She grew to like them over the course of the next two days, but her first reaction was "Yech!" She thought it was way too structured and Beatle-esque, and we thought it was a complete work of genius. By that time we'd been up for 20 hours and we thought it was the greatest thing!"

Andrew Gold, *What's Wrong With This Picture?,* 1976

"On that album, one of my functions was filtering Andrew's ideas. He's so musically capable, and has such a tremendous ear, he can think of a zillion different overdubs to do on everything. I love "Lonely Boy," I love the song, the way it

their mark, like The Innocence Mission and John Kilzer.

One also can't help but notice that in an industry filled with volatile and transient working arrangements, Asher's 16-year production history with Linda Ronstadt and 20 years of managing James Taylor represent unusually long-lived relationships.

Asher is a youthful-looking 45. Upon meeting him, one's eyes are drawn instantly to an impossibly red head of hair. His energy level is that of a man 25 years younger; even his speech races along at twice the normal pace. He has a pleasant British accent, tempered by many years of living and working in the States.

Like so many of their generation, Asher and a classmate, Gordon Waller, got their musical start in London coffeehouses, singing and playing Everly Brothers and similar songs, while still in their teens. They became known as Peter & Gordon, for a time touted as "the biggest thing to hit the American music scene since the Beatles."

Actually, the Beatles were in no small way responsible for launching Asher's production career. Asher first encountered them when his sister Jane reviewed a Beatles concert early in the Fab Four's career and got to meet (and almost marry) Paul McCartney. Not too long after that, Peter & Gordon's rendition of "A World Without Love"—a Lennon/McCartney song—became one of the duo's biggest hits. McCartney had played them an unfinished version of the song just about the same time Peter & Gordon won a record contract with EMI/England and were planning songs for their first session.

At the tender age of 24, Asher was asked by McCartney to head up the A&R department of the newly formed Beatles label, Apple Records.

It was a casual arrangement. Just after starting there, Asher "found" James Taylor through a mutual friend, guitarist Danny Kortchmar, who had been in a New York band called The Flying Machine with Taylor. When the band broke up, Taylor decided to move to London to seek his fame and fortune. Kortchmar gave him Asher's telephone number. Meeting with Asher in London, Taylor played a tape that knocked him out, and Asher said, "This is perfect. I'll sign you to this new label and make a record." (Unfortunately, and because of the lengthy legal disputes surrounding Apple, that record has been unavailable for years.)

The beginning of the '70s found Asher in California, continuing his production work with James Taylor, James' sister Kate Taylor, and others. Asher became a manager "by default," as he puts it, when he came to the U.S. with Taylor and they weren't sure who to hire for the job. He understood the Wall Street concept of contrariness long before it had a name, and based his management style on what he'd seen done *wrong* with Peter & Gordon's career. Two decades later, he's still working with James Taylor; Asher even sent him a gold watch on

Producer's Tip:

IF A RECORDING SOUNDS GOOD, don't start putting overdubs on just for the sake of something to do, thinking, "Maybe it would sound better with 10,000 xylophones." On *Sweet Baby James* I started to worry: "God, it's kind of simple-sounding, and we didn't spend much money, and we got it done in two weeks. What did we do wrong?" Don't worry about that. It's much easier to spoil a record by putting stuff on than by leaving stuff off.

—*Peter Asher*

WHEN YOU'RE MIKING ELECTRIC GUITAR, try *everything*. Mike the amp close, mike the amp far away, take the guitar direct, put the guitar through two different amps, try a Leslie—everything. I remember doing the Maniacs record, I think it might be on "What's The Matter Here?", we were trying desperately to get an electric guitar sound. It sounded great when we were doing it. The guitarist was sitting in the control room with us, and then we played it back and it didn't sound so good. I realized one thing I was hearing was the actual acoustic *zigga, zigga, zigga* of him hitting the strings. And I went, "That's what I'm missing!" So we miked that, and recorded it again. We had to turn the monitors way down and work on headphones, 'cause that mike would have fed back like mad. You had to turn it way up to get the sound. But it's brilliant, sounds like a little mandolin on top of the guitar. The answer is just to be open-minded.

—Peter Asher

their "anniversary."

When first offered the job of managing Linda Ronstadt, whom he'd seen in New York years before ("I was completely enthralled," he remembers), Asher actually had to decline because his client load would have been too great. He started managing her before it was determined that he would produce her. They collaborated in 1974 on *Heart Like A Wheel*, liked the results, and have been working together in the studio ever since.

Asher's latest Ronstadt opus, *Cry Like A Rainstorm, Howl Like The Wind*, is more a "production" in the true sense of the word than her series of records with the Nelson Riddle Orchestra. Recorded at George Lucas' Skywalker Ranch facility in Northern California by esteemed engineer George Massenburg, the release also features the work of the Skywalker Symphony Orchestra, Oakland Interfaith Gospel Choir, and the Tower Of Power horn section. Several cuts, including the title track, highlight vocalist Aaron Neville of the Neville Brothers, whom Asher considers "the foundation and backbone of New Orleans music." Ronstadt and Neville also took home Grammy Awards in February for Best Pop Vocal Performance by a Duo or Group for the track

"$\mathcal{S}ome$ *of the technical deficiencies are in themselves appealing."*

"Don't Know Much."

Since the early '80s, Asher has broadened his palette somewhat, working with 10,000 Maniacs (1989's *Blind Man's Zoo* and the album that "broke" the band, *In My Tribe*), Cher, and Diana Ross, and venturing into the Broadway scene with the *Pirates Of Penzance* cast album. Most recently, Asher completed tracks for a forthcoming album by Boy George's Culture Club.

Leafing through a copy of *Billboard* that's lying on a table, I see that *Cry Like A Rainstorm...* is 14 with a bullet, and then all of a sudden Asher bounds toward me from a doorway (through which I've seen him energetically pacing for the last ten minutes) and apologizes for the wait. A few moments later we're comfortably ensconced in a well-lit back room near a Yamaha baby grand, and the tape's rolling.

• • •

EQ: *I grew up listening to James Taylor and Linda Ronstadt. Now I find it ironic that such quintessential American music, played by quintessential American artists, was produced by an Englishman. Were you conscious of a jump from making English music to making American music?*

Peter Asher: I wasn't conscious of any jump at all. But you have to remember that everyone in England did nothing but listen to American records; we were American-obsessed, in the same way that America then became English-obsessed. The '60s were largely the result of everyone trying to sound as American as possible. I tend to make records the same way now, with the same general intentions, as I did then. A lot of other things have changed—the technology, of course, has changed vastly—but my overall methodology hasn't changed that much.

EQ: *Is modern technology ever an imposition?*

Asher: No, it doesn't impose itself. I think *you* use *it*. You have to make sure you're the boss. But it means that I can sketch out tunes on my computer at home. I use a Yamaha C1, which I love, because I'm an IBM man by nature. It does everything the Mac does, but you don't have all those extra boxes and wires because MIDI and SMPTE and everything goes straight in and out the back. I've used that for everything lately. I did the Cher tracks completely that way. I wrote everything at home on the computer, brought it in, laid it all down, put some real key-boards on it, then got a bass player and drummer in and had them play it proper-

started. There are a lot of parts, and there could have been a lot more! If anything, I tried to keep it more straightforward rock 'n' roll and a little less overdub festival, because he does have a lot of ideas to try."

Pirates Of Penzance, Broadway Cast Album, 1982

"There are two different *Pirates Of Penzance*: the cast album and one you've never heard. The show album we did like they do most Broadway shows, which is pretty much live, lots of takes, vast amounts of editing. This was predigital, or it would have been a lot easier. I basically had cassettes of all the different takes and would sit making bar notes where I wanted to make all the edits to get good versions. Then there's the movie soundtrack, which was never released as an album because they would have had to pay all the musicians again, and they couldn't be bothered. But it actually is a much better recording. That one we did pretty much one piece at a time, overdubbing separate parts."

10,000 Maniacs, *In My Tribe*, 1989

"Part of being a record producer is diplomacy. Even with studio musicians, you still want to get everyone on your side, get a good feeling going in the studio, and a certain sense of cooperation. Yelling at people definitely doesn't pay off. When you're dealing with a band, you're adding to that the fact that every band member has a say. With studio musicians I always ask, "What do you think would fit?" because [otherwise] they'll basically sit there and wait for you to tell them what to play. With a band, *they'll* tell *you* stuff. You spend a certain amount of time resolving conflicts between different ideas about how things should be done."

ly, and there was the track. It was a breeze.
EQ: *Do you have any favorite equipment?*
Asher: Neumann U67s are probably the
best microphones in the world, for almost
everything, but vocals in particular. I like

the new Bruel & Kjær mikes. I
used one on Natalie Marchant
of 10,000 Maniacs, the sort of
thin, pointy one, the B&K
4006. The Neumann M49s
that George Massenburg usu-
ally uses as overheads for or-
chestras are amazing, and the
best equalizers and compres-
sors are actually George's. He's
got a new limiter that there's
only one of at the moment;
the queue of people wanting
to buy one is huge. Other than
that, all the usual stuff, stan-
dard things everybody uses. I
like the new Akai S1000 sam-
pler. I use the Forat F16 for
drums a lot. It might be the
best drum sampling machine
around. Plus I like a lot of old
tube stuff.

EQ: *With the exception of the
new album, Linda Ronstadt's records rarely
have more than two songs by the same per-
son. Did that cause you any problems get-
ting an album to be cohesive or getting con-
sistency from album to album in defining a
Ronstadt style?*

"The '60s

**were largely
the result of
everyone try-
ing to sound
as American
as possible."**

Asher: No, because part of
her genius is that she
picks songs that fit some-
thing she wants to say. So
by the time she's picked
them all, and they're all
right for her—which
doesn't mean we don't
sometimes try songs and
throw them out—they fit
together, because they all
fit with what she's trying
to do.

EQ: *There's never been
heavy use of synthesizers
on Ronstadt albums. Does
she dislike them?*

Asher: She's not very keen
on them. I've always tried
them where they would
be appropriate. There's ac-
tually synth on most
tracks on this album, but

it's mostly there to accentuate or fill in
other parts. You'd notice if it wasn't there.
Linda had an idea of the way that synths
sound on synth records she doesn't like.
She didn't want any synths on this record.
But I said, "Let's try, because this guy
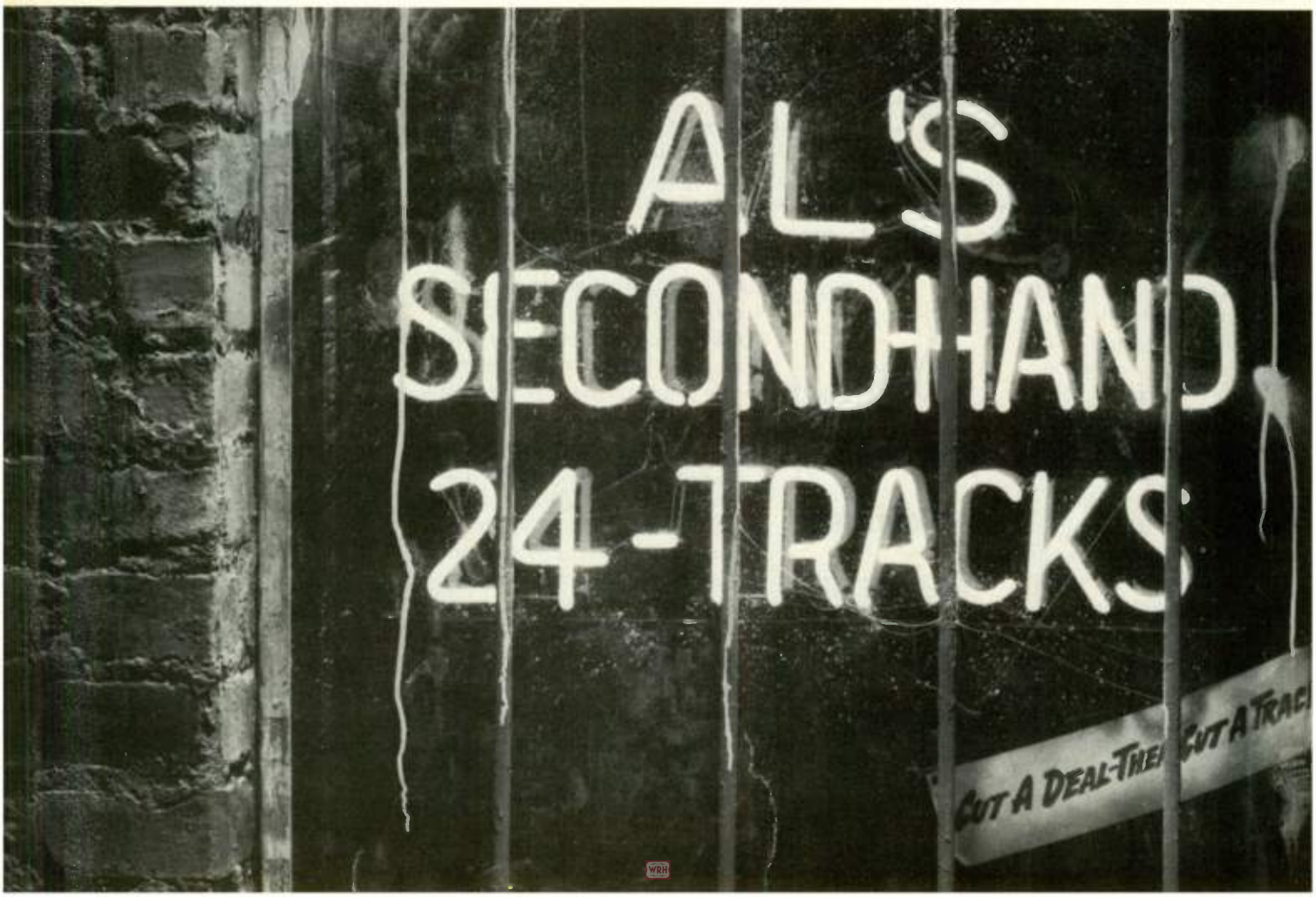
Robbie Buchanan is very good and he has a lot of sounds I think you'll like, and I don't think it'll sound like synthesizers when it's done."

EQ: *Did you have that in mind from the start, or was it a matter of recording with the orchestra and finding they needed something extra?*

Asher: I had it in mind from the very start.

EQ: *Did she know that?*

Asher: No. [Laughs] I think I said, "I want to try some synths," and she went, "Well, okay," wrinkling her nose as only she can. But one reason I'm sure we get on is that we're both ready to admit when we're wrong. When she heard what Robbie's capable of doing, she realized they did work. We've had similar arguments the other way around, of course, when I'm sure something is not going to work and she's completely right.

EQ: *When she first told you that she wanted to do the Nelson Riddle records, were you surprised or did you know that her heart was leaning in that direction?*

Asher: It was something she'd been talking about for years, so I knew it was on the horizon. But when the time actually came, of course, it was a disconcerting prospect. I

had no idea how well it would do. I had to learn a lot about the songs, and about Nelson. It took a great deal of homework. It was a big orchestra with Linda singing live, and there I was following a score and making notes bar by bar, and it's a whole other kind of panic from making records one layer at a time in the way we were accustomed.

It was intimidating to me in the sense that Nelson is who he is. Neither [Linda nor I] read music, and we both worried that he would be used to a whole other level of expertise and slickness that we couldn't provide. But he turned out to be totally cooperative and charming and helpful in every respect—which didn't mean he couldn't be just terrifying on a date. He had musicians absolutely trembling! He was hard as nails, but brilliant.

"*It would have been easy for 10,000 Maniacs to go, 'Oh, no, we don't want some old fart producer. . . .*"

EQ: *Because people knew a Nelson Riddle sound, was there pressure on you to match or better it?*

Asher: Well, our intention was to better it. Of course, there are some things about those old sounds that you fall in love with and can't match. Some of the technical deficiencies are in themselves appealing. Also, at that time everyone's expertise at recording a large orchestra with a few microphones was higher than it is now. But in my view it's the best-sounding orchestra I've ever heard.

EQ: *How was the orchestra recorded?*

Asher: We used some spot mikes, but we largely relied on a pair of stereo overhead mikes, [Neumann] M49s as I recall, and that is largely what you hear on the record. We used the individual mikes to boost up certain parts in certain places, to get a little

extra celli when we needed them for richness, for example.

EQ: *Why did you record* Cry Like A Rainstorm... *at Skywalker Ranch?*

Asher: It was an acoustic decision, largely. Linda had been up there because George Lucas is a friend of hers. She went to a party and they had a band playing in that room and she loved the acoustics. She came back and told George Massenburg and me that we should go up and listen to it. So we did, and we agreed with her.

EQ: *Could you describe it?*

Asher: It's huge, monstrously huge! The acoustics are totally variable. It can be all hard—masonry, not wood, which I like better—or all soft. So you can go from a huge, giant, perfect reverb to a relatively dead sound, but still have the space of this very big room.

EQ: *You mean they have all those environments within that space, or they use treatments?*

Asher: It's one giant room. You press a million buttons and everything [panels on the walls and ceiling] slides back and forth. George figured out what we'd have to do technically to make it work for us to record there, and we did. It's also a nice place. I like the idea of being in a residential studio. They have very nice accommodations. You can walk to the studio every day across beautiful California scenery.

EQ: *Equipment-wise, what's it outfitted with?*

Asher: A Neve V Series console [with GML automation], and we brought in a couple of [32-track] Mitsubishi digies, 860s or 850s, whatever they are. I always get my numbers in a muddle. And there's an awful lot of stuff George brought with him.

EQ: *Was it also mixed digitally?*

Asher: Yes. [On the] high-sampling rate Mitsubishi X-86 [2-track machine].

EQ: *When you did* Pirates Of Penzance, *did you work on a stage to try to retain the stereo integrity of a performance?*

Asher: I think we were in the old, big Columbia studio in New York that's now

> "*A little too much reverence gets attached to certain versions of songs.*"

torn down. I tried to recapture some stuff from the show. I know we had several mikes strung around the place and a pair of stereo mikes overall. I would assign each singer to a microphone, so that by the time they got to their solo song, they'd be not on the mike, but six feet back. But I knew which one they'd be on, so I could at least regulate the level.

EQ: *Did you use processing then for the reverb?*

Asher: Both. Some room and some reverb.

EQ: *What was it like to work on the 10,000 Maniacs and Cher projects?*

Asher: The A&R man at Elektra, Howard Thompson, suggested me for the Maniacs. I had never heard them. I listened to their early independent albums, and I loved them. So he set up a meeting with Natalie, and we got on well. The first thing she asked me was if I smoked. I said, "No," and I was halfway home already. [Laughs]

Linda's the same way—she won't let anyone in the studio who smokes.

I was very pleased, because it would have been easy for them to go, "Oh, no, we don't want him. We don't want some old fart producer, we want some new, young, groovy producer because we're alternative." They didn't take that attitude at all. Making the record was really a pleasure. I like everyone in the band and we had a good time doing it.

Production for Cher was quite different from working with the Maniacs, where they're there all the time, they all have strong ideas about everything, they all participate all the time, and there's a lot of juggling and diplomacy. The first production meeting I had with Cher consisted of a phone call from her secretary saying the demo was fine. So I went ahead and made the record and she came to sing. I was very much on my own.

EQ: *Let's talk about the Maniacs remake of "Peace Train." In many people's minds, there's only one version of "Peace Train," the Cat Stevens version...*

Asher: Yes, but that applies to all the records Linda ever did. There's only one version of "Heat Wave," there's only one version of "That'll Be The Day." Buddy Holly's version of "That'll Be The Day" is still the original. Martha Reeves' version of "Heat Wave" is still the original. But I think if you have an interesting way of doing a song and you enjoy singing it, you should go ahead and do it. There's always going to be someone who goes, "How dare you touch that classic!" Screw them. If you want to sing it, sing it, and if people don't like it, that's okay, too. But if they do, that's great. And better still, some people may hear the song who have never heard it before. I think a little too much reverence gets attached to certain versions of songs. You should do any song you want to.

EQ: *Could you tell us about the first single you ever produced, the Paul Jones single, "And The Sun Will Shine"?*

Asher: It's really good, actually. I listened to it the other day. It's a Bee Gees song. Paul Jones was a very good singer, to this day sings great, and works all the time in England. He used to be the lead singer of Manfred Mann. He asked me if I would produce him, which was very bold because I'd never produced a record before, but I'd been very involved in the production of Peter & Gordon's latest stuff, and he'd watched me work. I put together a rhythm section which ended up being relatively star-studded, and the record has become a kind of classic now, because Paul McCartney played drums, Jeff Beck played guitar, Paul Samwell-Smith from the Yardbirds played the bass, and Nicky Hopkins played the piano. It sounded great. And then I put a lot of stuff on it, an orchestra and a choir. It was a minor hit in England.

EQ: *If you could pinpoint one difference between producing 20 years ago and producing now, what would it be?*

Asher: There are more similarities than differences. The difference, of course, is the technology. There are so many things you can do now that you couldn't do then. The possibilities are endless. And it's fascinating, I love it all. In the end, though, you're still down to finding a good song and getting a good performance. •
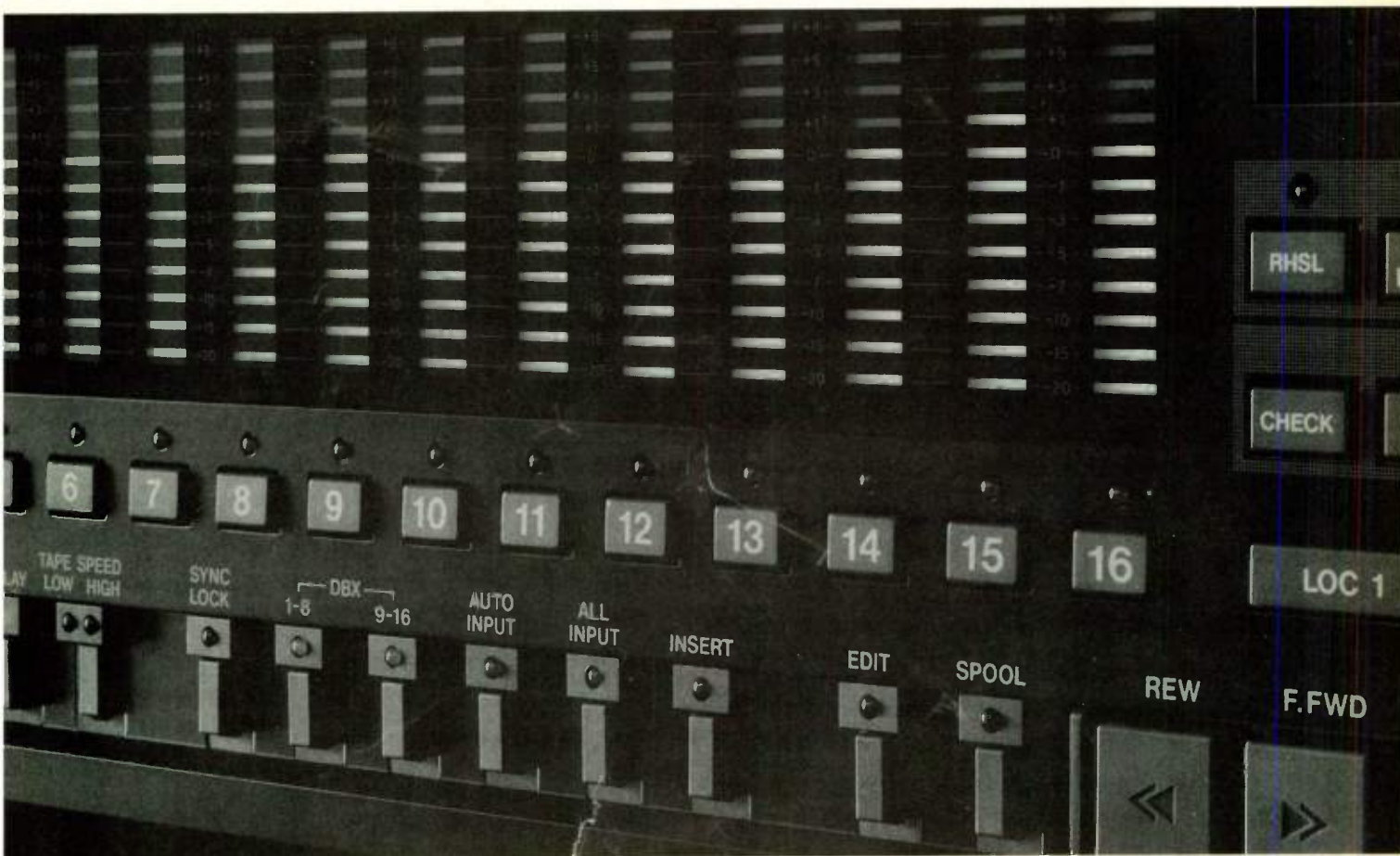
BASIC TRACKS

Preparing Drums For The Session

BY TOM LUBIN

THE STARTING POINT FOR MOST GROUP SESSIONS is laying the bed tracks. And the starting point for most bed tracks is getting a good drum sound. Some ancient musical mythology that has been passed down suggests that this process must take an inordinate amount of time—from many hours, to days. So today we'll look at getting good drum sounds.

Losing Your Head. Treat drumheads the way you treat acoustic guitar strings: Change them as often as you can afford to! At the least, put new heads on the kit before each session. More specifically, on the day before a session, take the following steps: (1) Take off the old heads. (2) Lightly sand the edges of the shells that touch the heads to remove splinters and the like. (3) Put on fresh heads and crank them tight. (4) Press on the heads to stretch them, then tighten them again. (5) Let the drums sit overnight and fine-tune them in the studio the next day.

Who Says Drummers Can't Hold A Tune? A good drummer can get a great drum sound on just about any set of drums—provided they are tuned properly. The best tuning for a given drum is usually the point at which it projects the most inherent resonance. Subjective, but true.

To insure proper tone, generally you want an even tension at all the lugs. Most drummers check the tension at all the lugs by lightly tapping the head near each lug post. Some drummers use drum-key torque wrenches that get it close, but the fine-tuning ultimately still comes down to tap, tap, turn; tap, tap, turn. After you complete the first tuning pass, give the heads a few good whacks to settle them in across the shell lips. Then tap, tap some more.

Remember, a drumhead undergoes a tiny change each time it is struck. If you value consistency, record some reference tracks at the onset of the session (strike each drum repeatedly while recording to a unique track). Then periodically tune your drums during the session, and compare them with the reference tracks.

If you decide to go for different drum sounds on different cuts (a subjective call), radical tuning alterations are not the solution. This refers back to our earlier premise that each drum has only one point where it resonates optimally. The best ways to establish a different character from track to track include signal processing and using a different drum set.

Quiet On The Set. It's also important to eliminate a drum set's hardware noise. Use silicon spray to eliminate squeaks in the pedals. Tighten down hardware that rattles. If any fittings move that can't be tightened or removed, use rubber cement or rubber washers to prevent them from shaking.

Also, make sure the surface *under* the kit doesn't move. I usually like a hardwood floor with a small rug between the kick drum, hi-hat, and throne. If the floor is carpeted, get a couple of 4' X 8' sheets of 5/8" plywood or particle board. Lay these sheets flat on the carpet so they don't vibrate, and keep all the mike-stand bases off the wood so they don't shake. Avoid setting up the drums in a small booth, because this creates too many phase problems.

Are Two Heads Better Than One? Another subjective matter is whether to use or remove the bottom heads on the toms. The top head is used largely for concussion, while the bottom is for resonance. The single-headed approach is really a quick fix, yielding more specific pitch and less resonance. If you want to do it right, first tune the bottom head to the approximate pitch while striking it. The top head then is tuned to the drummer's taste in terms of rebound. This process requires more time, but yields better results.

As for the kick drum, it's more or less standard operating procedure to remove the front head, unless you're going for an extremely resonant sound. Single-headed kicks are more punchy. (Many drummers will have a front kick-drum head with a big hole in the center, but what's left of the front head really just keeps the hardware from rattling!)

Most of the drummers I know put a pillow or a blanket inside the kick drum to damp the shell's vibration. The best way to take care of all the rumbles is to line the inside of the drum shell all the way around with a 3" piece of closed-cell foam rubber, the same width as the shell. (Pillows generally yield more thud; foam gives more resonance.)

I completely pull out all the internal muffles on all the drums to get a big open sound. The internal muffler is used primarily for live work and while it can be useful, it puts pressure unevenly on only one side of the head, making it sustain and dissipate with some nasty overtones. Better to use small pieces of gaffer's tape and bits of cotton or gauze to create even dampening. "Dead Ringers" also work well (rubber gaskets with adhesive backs that attach right to the head to dampen it).

Dozens of interesting modifications and variations apply to drum sounds. Ringo Starr used to bang his snare through a towel. Some drummers will tape anything to their snares to get a different sound—wallets, key cases, sanitary pads, you name it. Here's my favorite trick for getting rid of overtones for a dry sound: Get an old snare drumhead and cut off the ceramic hoop. Then cut out a 5" hole inside the circle so you end up with a ring

> It's Possible To Record A Great Drum Sound On Just About Any Kit—Providing The Drums Are Tuned Properly. Here Are The Fundamentals.

MIDI & COMPUTERS

Ready, Set, Automate

BY MIKE COLLINS

IN LAST ISSUE'S COLUMN WE discussed the Macintosh-based Hyper-Card, HyperTalk, and HyperMIDI software—the latter is a language extension for HyperTalk which incorporates MIDI communication functions. We also promised to describe how to write your own HyperMIDI application that makes it possible to take an "electronic snapshot" of any and all MIDI gear used in a given session, to be used for future recall.

Well, the editors at *EQ* have decided to go one step further: Just follow the instructions at the end of this column to receive your own free copy of *EQ SysEx Snapshot* on disk. In the meantime, let's take a look at the HyperMIDI programming process.

About The Program. Like most HyperCard-based applications, the basic element of *EQ SysEx Snapshot* is the stack. Each card in the stack corresponds to the SysEx data "snapshot" from a given session. Each button on the stack has its own script dealing with a given MIDI device, such as a signal processor or keyboard. The program number for a given device is displayed inside the corresponding button.

Each device icon also has a STORE and RECALL button. STORE retrieves the data corresponding to the current program from the device and saves it with the session's card. RECALL brings up the session data for that device, then transmits it back to the unit's internal memory. A PROGRAM CHANGE command also is sent to the unit, so the new data becomes active.

The stack also provides STORE ALL and RECALL ALL buttons to automate the process of capturing and dispersing session data for all your MIDI devices. When the dumps are captured, they are stored within the stack itself. You simply use one card for each project, and go to that card when you want to recall the project's settings. (Or you could call up a card from another session, select a killer setting for one particular MIDI device, and send that patch to that instrument alone.)

There is even a provision for a MIDI patchbay—highly desirable since the MIDI signal flow must be routed properly (and usually differently) for each device. (Remember, while most computer MIDI inputs are at a premium, many devices require a "handshaking" loop with the computer.) Each device in your setup can be associated with a program number on a programmable MIDI patchbay: When the program sends or receives data for a certain unit, the corresponding PROGRAM CHANGE command is sent to the patchbay to insure proper routing.

Clicking on the NOTES icon at the top left of the stack reveals a text field for entering session notes. A text field along the top of the window is provided for the name of the particular recording project. If you want to add your own buttons or alter the appearance of the card, just do so in HyperCard using standard techniques.

What You'll Need. First, you'll need a copy of HyperMIDI 2.0 from EarLevel Engineering (21810 Barbara St., Torrance, CA 90503; (213) 316-2939; $195.00). This collection of external commands must be installed in your stack as Macintosh Code Resources, using a utility program such as ResCopy or ResEdit.

The next thing you'll need is a desire to get a few bits and bytes of SYSTEM EXCLUSIVE code under your fingernails. [*Ed. Note: Stay tuned for more on SysEx in future issues; in the meantime, you might want to check out* The MIDI System Exclusive Book *and* The MIDI Resource Book, *both by Steve De Furia and Joe Scacciaferro, and published by Ferro Technologies of Belleville, NJ.*]

We'll give you the basic framework of the program, with a completely functional button for a Yamaha SPX90 or SPX90-II. You can use that button's script as a template to pattern additional buttons for additional MIDI gear. The disk also includes SysEx listings for Yamaha SPX1000 and REV5, Lexicon PCM70, and the TC Electronics TC 2290.

You'll need the SysEx bulk dump requests for any other units you wish to incorporate. If they're not included in the operator's manual, contact the manufacturer or distributor and ask for this information.

Free Software! *EQ SysEx Snapshot* is not the end-all in studio automation. In actuality, few products exist which address today's MIDI automation needs. For interested parties, however, it will provide a starting point from which MIDI applications for your own setup can be customized. It is much easier to write a Macintosh program using HyperCard than in a traditional programming language such as Pascal. I hope this will encourage more studio engineers to create custom programs to control their studio's MIDI gear.

To get your copy of *EQ SysEx Snapshot,* send a Macintosh-formatted 800K floppy disk and a self-addressed, sufficiently stamped envelope to: *EQ SysEx Snapshot*, 20085 Stevens Creek Blvd., Cupertino, CA 95014. •

Mike Collins is a London-based freelance recording engineer, and a contributor to Britain's Studio Sound *magazine.*

Automate The Session Data For Your MIDI Gear—With A Single Click Of The Mouse

STUDIO CLINIC

Taking The Tape Path To New Heights

BY DR. RICHIE MOORE

PROBABLY *THE* MOST IMPORTANT FACTOR in tape path maintenance is the height of the tape path, in relationship to the magnetic heads. The head blocks of all tape machines have adjustments—including azimuth, wrap, tilt (zenith), and height—for the heads. Usually there are fixed guides within the head block that reduce vibration during tape movement, and maintain the tape's height. The manufacturer uses a test jig to set the height of the fixed guides, using very high tolerance, machined metal blocks on a piece of lab-milled granite slab (flat to a tolerance of 0.0001"). Once the head guides are set, the heads then can be adjusted for proper height for the tape path.

In the field, head alignment usually is just one adjustment—azimuth (phase). Improper azimuth alignment results most obviously in poor high-frequency response. You can adjust azimuth with the help of a *standard reference tape*. Manufactured by Standard Tape Laboratories (STL) and Magnetic Reference Laboratories (MRL), these tapes are pre-recorded with tones; in most cases, heads are adjusted to provide maximum output with a variety of tones. Over many years, however, we have learned that the amount of variation in the phase of these test tapes can be considerable. Therefore, whenever a tape comes to you from a studio other than your own, the azimuth should be set to the tape you're using for maximum high-frequency output, checked on an oscilloscope.

The other head adjustments (wrap, tilt, and height) typically can't be performed for every session, because of studio time constraints. So here's a simple test for checking your head height adjustment:

1) Record a 1kHz tone, followed by 10kHz, on *all* tracks in one direction, at 0VU.

2) Play back the tape and write down the VU level readings for each track.

3) Turn the tape over and play it backwards. Write down the new levels.

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

Yes, Tape Adjustment Can Be Tedious. But You've Got To Do It. Here's The Right Way.

4) Check to see that the playback levels match ±1.5dB on all channels. If there is greater fluctuation, call a good technician.

Now that we've established that tape path height is dictated by the head block guides, let's work out from the center, and look at the rest of the tape path.

Shine a flashlight on the oxide (shiny) side of the tape to check for the following conditions: The rollers and guides on the tension arms should be the same height as the head guides. When the tape passes by the guides, there shouldn't be any visible ripple on the tape, and the top and bottom of the tape should not be curled or warped in any way. The tape should be positioned so it is equally spaced between the top and bottom of the guide rollers.

Moving out a bit further, we meet the reel turntables. Their height should be set so that the tape, as it's coming off, does not rub on the top or the bottom of a reel flange. Test this at normal *and* high speeds. Regularly check the reel turntable height; it often is out of adjustment. Reels are dropped on the turntables and people tend to rest objects on them. Also, during transport, the set screws that hold the reel height can come loose. (Beware of plastic or damaged reels—they often are warped and will rub the tape, regardless of your deck's alignment.)

In addition to being the same height, the turntable and guide rollers also must be parallel. This is why it's important to avoid applying any weight or pressure that might bend these components.

Always use a new roll of tape to examine the condition of the tape path, to assure that you are not just looking at a bad roll or a bent reel. Once the transport height is correct, the heads adjusted, and the transport cleaned and demagged, you are ready to move on to the electronics alignment and setup.

Next time we'll tackle reference levels on magnetic tape and their importance (an area with a proven track record of confusion). Until then, keep up the preventive maintenance.

• • •

Capstan Cautions. Last issue we discussed an extremely important procedure—the cleaning of the tape path. I wanted to clarify a point or two regarding cap-

A typical test jig, shown here on a Tascam ATR-80 24-track recorder.

BRIAN HAGIWARA

Getting high before takeoff.

If the thought of flying doesn't scare you, how about the thought of flying with a pilot who's addicted to cocaine?

According to a series of articles by a Pittsburgh newspaper, drug addiction among airline pilots is not at all uncommon.

The newspaper's series was the result of a phone tip from a frustrated doctor.

He told of a pilot who'd been rushed to the hospital near death from an overdose of cocaine.

A survey of the area's hospitals proved this was not an isolated incident.

Upon further investigation, reporters learned that screening flight personnel for drug use was forbidden. And that the FAA medical exams didn't mandate blood and urine analyses so drug use couldn't be detected.

They learned scores of pilots were receiving treatment for drug abuse.

The newspaper's story brought the problem to the attention of the public.

The public was outraged. And demanded immediate reform.

As a result, several reforms were initiated. Among the first, was the development of more extensive FAA medical exams.

Reforms that may never have occurred if it hadn't been for a couple of reporters pursuing a story.

To learn more about the role of a free press and how it protects your rights, call the First Amendment Center at 1-800-542-1600.

If the press didn't tell us, who would?

Neumann U87 Studio Condenser Microphone

By Klaus Heyne

Illustration by Rick Eberly

F THEY GAVE OUT Grammys for microphones, the Neumann U87 would have a mantle full of them. This venerated condenser microphone, in continuous production since 1967, has become legendary through its use in top studios around the world.

It also is an ideal subject for our "How It Works" investigation. The U87 is a typical all-purpose studio condenser mike that works in a wide range of vocal and instrument applications.

1. Head Grille. The grille protects the fragile capsule, and shields it from hum. It does not protect the capsule from high sound pressure, saliva, dirt, or

moisture. (See the accompanying "Golden Rules" for long-life tips.)

Notice the wedge-shaped screen: It's slanted to cut down on sound bouncing back and forth between capsule and screen (standing waves).

2. Capsule. Consists of one diaphragm system—or two, back-to-back, as with the U87. Each system has a thin, metal-covered plastic membrane which, when excited by soundwaves, vibrates closely in front of a metal backplate. As sound vibrations constantly change the distance between moving diaphragm and fixed backplate, the electric capacitance created by these two metal plates changes accordingly.

By connecting this "capacitor" (old term: *condenser*) to a DC (polarization) voltage, the changes—with some help of additional circuitry in the amp-section of the mike—translate to a small AC or output voltage. This output eventually is amplified by the mike preamp of the mixing console to usable (line) levels.

3. Directional Pattern Switch. This determines the mike's angle of sensitivity to oncoming sound. This particular model offers three distinct pickup patterns:

a) Cardioid. The microphone is sensitive only to an angle of about 30° to each side from the head-on position 0° (identified by the firm's logo).

b) Figure Eight (Bi-polar). It's simply a double cardioid. The cardioid pattern is electrically combined with the same pickup pattern from the rear of the mike, but the rear cardioid is phase-reversed 180° to maintain two distinct cardioid patterns when added together.

c) Omni. The omni pattern features the same combination of two cardioids as the Figure Eight, except that both cardioids are combined *in* phase, to achieve 360° sensitivity—equal on all sides of the mike.

4. -10dB Switch. This lowers the capsule (not amp!) output voltage by 10dB, to protect the mike circuitry from electrical distortion in high sound-pressure conditions. Contrary to common belief, activating this switch does not suppress mechanical diaphragm distortion, which often occurs sooner than amp distortion in extremely loud situations.

In addition, a substantially decreased signal-to-noise ratio makes this an "emergency only" switch.

Golden Rules Of Preventative Maintenance

THE THREE most important steps to guarantee a long life and undiminished performance from your condenser mike are easy to perform and cost almost nothing.

1. Stretch your hand—from the tip of your thumb to the tip of your pinkie. That's the minimum distance to any sound source a microphone capsule can be without getting mechanically "tired" and eventually losing tension.

2. Always use a stocking screen for vocal work: The head grille over your capsule offers little protection against saliva spray

onto the diaphragm. Saliva is starch, starch is sugar, and sugar is sticky. Since the capsule has an electrostatic charge (polarization voltage), dust particles fly straight to the diaphragm, from as far as 6' away. These bits of debris become lodged on the now-sticky membrane; soon the movement of the diaphragm is slowed down by the weight of the dirt. High frequencies go first. Eventually so much dust collects, it shorts to the backplate of the capsule, resulting in thundering discharges. When the bitter end is near, the output level of the mike starts to

cut out completely. Not a pretty picture.

3. To minimize the static attraction of dust to the capsule, pull a plastic bag (from your grocer's produce department) over your mike during overnight breaks in recording sessions, when you need to leave it on the stand. When the session is over, wrap the mike in the same bag before transporting or storing it in its case. A plastic bag is also effective and cheap protection against moisture build-up in the capsule (for example, when you are traveling in rainy weather).

—*Klaus Heyne*

5. Low-End Roll-Off Switch. This dramatically rolls off low frequency response—as much as -12dB at 40Hz. Like all severe cut filters, it affects the overall clarity of the mike somewhat.

6. "Amplifier" Compartment. Actually, amplifier is a misnomer, since the capsule output (microphone level) is already hot enough and doesn't need further amplification. Instead, other vital functions are performed on this circuit board: Rough supply voltage coming from the mike preamp or mixing board (see #9) is refined and boosted here, before being sent up to the capsule as polarization voltage.

7. Field Effect Transistor (FET). The "amp" also converts an extremely high capsule impedance of more than 1,000 meg-(million!) ohms down to a few thousand with vital help from a field effect transistor. Until the mid-'60s, vacuum tubes performed this impedance link-up; hence the term "tube" mike.

8. Output Transformer. The transformer further lowers impedance, to about

200 ohms at the output of the mike. That's a typical mike impedance level, appropriate for most 600-ohm console mike inputs.

9. 3-Pin XLR Connector. Normally, a microphone such as the U87 is plugged into a standard 2-conductor shielded mike cable, and connected to a phantom power-equipped console input. A 48V DC supply voltage, generated in the console, travels like an unnoticeable phantom to the mike, using the same two cable conductors that eventually carry the sound signal from the mike back to the console.

10a) Battery Switch. When phantom power is not available, battery operation is an alternative powering method, yielding about 200 hours of recording per set of batteries. (The option of battery power is not found on most condensers.)

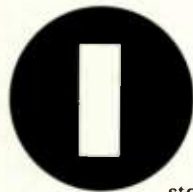**10b) Battery Holder.** In the U87, this holds two 22.5V batteries.

10c) Battery Status Indicator. When the needle on the battery indicator at the rear of the mike leaves the red field, it's time for battery replacement. •

FILM & VIDEO SOUND

Editing For Film: Mag Or Multitrack?

BY LARRY BLAKE

I N A COLUMN I WROTE for *Recording Engineer/Producer* a few years ago, I expounded on the relative merits of 35mm mag sound editing vs. multitrack sound editing. Editing "mag stock"—which resembles 35mm film, although it has a "coating" of magnetic oxide running along it, to record and play back sounds—is a very mechanical process: Dialog, sound effects, backgrounds, and other "tracks" all are kept conveniently on separate "units" (1,000-foot reels). Editing together these different components involves cutting and splicing appropriate sections of units, while running in mechanical sync with the film picture—sprocket by sprocket. What could be easier?

Multitrack tape, on the other hand, seemed cumbersome and prone to problems: Recording effects or dialog onto multitrack requires electronic synchronization to film or a video worktape (rather than straightforward, sprocketed mag synchronizers), complicated time code (rather than frames you can hold in your hand), and so forth. These "complications," along with some other advantages of mag editing, made mag editing the clear winner—or so I thought. And that's what I wrote.

The column was greeted with raspberries from a friend of mine who is a sound editor and mixer, and also co-developed one of the leading synchronizer/editing systems. "You'll see one day," he said, and implored, "Try multitrack." I said I would, and that if my opinion changed, I would eat crow in a future column.

Pass the crow.

I now realize that many of the "problems" with multitrack editing have nothing to do with multitrack editing itself. Rather, they often are a result of an editor's ignorance or lack of editing ability—something that can put a dent in *any* form of editing. Improper use of time code or a botched film-to-tape transfer (which should be punishable by law) can cause headaches.

My turning point came in December 1988, when I began working as sound editor and re-recording mixer on the film *sex, lies, and videotape* ("*slv*"). Working with a 24-track machine (2"), time-coded 1/4" production tapes, and a video workprint of the film, I found a lot to

Larry Blake is a sound editor and re-recording mixer with Weddington Productions in North Hollywood.

like about multitrack editing. The ability to enter small offsets in the synchronizer at play speed, while looking at the picture, made it easy for us to achieve the elusive lip sync. It was really fun for director Steven Soderbergh and me to tweak sync ("what do you think, a half frame early?") and watch the dialog slide into shape.

Then I used a sampler to loop small segments of "fill" (ambient room tone, as explained last issue). This let me add five-second room-tone "handles" with ease to the beginning and end of each take—and that's something mag editors just don't have the time to do. When recording fills on the multitrack, it was easy to adjust their levels. This was helpful when attaching handles whose background levels changed during a take. And then there's the sound quality issue.

A 24-track tape recorder sounds far superior to the average 35mm mag machine. For *slv*, we used Dolby SR to encode the multitrack, which resulted in a pristine, clean recording. It is possible to cut dialog on SR-encoded mag, but because the picture editor's worktrack (the mag that accompanies the workprint) is not encoded, it's tough for the sound editor to compare it against the new, SR-encoded "reprint." And since few mag editing setups have noise reduction decoding capabilities, it's hard to judge what is really on the reprint. During multitrack editing, however, you're either listening on input, through the machine, or sync, with the SR-encoded track being decoded.

Mag editing *is* superior to multitrack editing when cleaning up small, bothersome noises that are closely interspersed among words, or when editing tiny fragments of words. That's because mag can be listened to, in sync to picture, at extremely slow speeds—something that is just not possible in your standard multitrack editing system.

As far as I'm concerned, the *main* reason to use mag film for editing and mixing feature films is its ability to conform to changes in the picture edit. It is not uncommon for sound editors to receive "change notes" from the picture department detailing 100 changes within a 10-minute reel, to be made overnight! It is an easy matter on mag film to scramble the order of units, thus allowing producers, directors, and picture editors to make changes up to the last moment. Contrast this to episodic television, where multitrack editing has made its greatest inroads: In TV, the picture edit usually is "locked" once the show gets to the sound editors.

With multitrack editing, the way to conform to picture changes is to use a second synced multitrack, to which you can transfer the tracks in a rearranged order. Extra tape generations may mean a loss of sonic fidelity, however, and an extra machine means extra expense.

Interestingly, just like rickety old mag film, those new high-tech digital audio editing systems (like the AMS AudioFile and NED PostPro) make it easy to slide around and rearrange dialog, sound effects, and other elements. Next issue we'll learn how disk-based editing systems offer their own set of advantages, and in some ways combine the best of multitrack and mag. •

Contrary To Traditional Belief, Multitrack Tape Can Make Sense For Film Editing
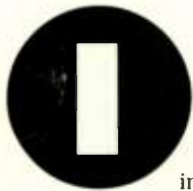
THE BIG PICTURE

Editing Suite Or Production House?

BY ROBERT WAIT

IF YOU ALREADY OWN or have access to an audio studio, and decide it's time to incorporate visual technology, the first matter to settle is which end of the film/video business you want to get into.

For many, making your studio ready to handle a scoring session requires taking relatively small technical and financial steps. However, setting up a video editing suite or film/video production facility could be quite a big leap. Let's look briefly at what's required for each.

Setup For Scoring Music Or Sound Effects To Picture. In any scenario that links audio and video, the first concept to tackle is how to sync the two formats.

Let's assume your audio studio already uses SMPTE time code to sync a sequencer to a multitrack recorder. SMPTE time code comes from the video world—via the Society of Motion Picture and Television Engineers—so any studio already using SMPTE as its sync signal is ready for the next step.

When integrating audio tape with video, you'll need a transport synchronizer (they start at under $1,000) and a multitrack recorder that can accept external sync. [*Ed. Note: We'll look closely at synchronizing audio to video in an upcoming issue of* EQ.] You'll also need a 1/2" or 3/4" video deck that can play a SMPTE signal on one of its audio tracks while playing the work track (the original recorded dialog for the video) on its second track. A quality 1/2" VHS stereo VCR will do the trick, although a 3/4" VCR is the more professional choice (you can buy one for as low as $3,000). Whatever you pick, the least expensive approach involves setting up the video deck as the time-code master to which all other devices in your studio will slave. Cue points (for laying in effects and music) will be read from the code striped on the "workprint" videotape. As a result, you'll be able to give your client a finished track with all cues in sync.

If your studio is not yet SMPTE-controlled, working with video will require you to sync "on the fly"—not an impossible task, but a time-consuming one. Syncing on the fly also implies that a sound or music editor will have to do additional work to accurately cut in the effects or music you've created.

Setup For Video Editing. Video editing suites come in two forms: *off-line* and *on-line*. A simple off-line editing bay has only the most basic gear necessary for editing video. This means one video recorder, one video playback deck, and one editing console that controls these two decks. Editing video is not a process of physically cutting and splicing videotape, but rather one of re-recording selected sections of footage from the original *dailies* (raw footage from each shooting day) to a *master* editing tape. Working in an off-line room, which is much less expensive to rent than an on-line room, allows you to spend time making the cuts exactly as you want on your *temp master* tape.

This accomplished, you move to the on-line room, where you'll turn your temp master into a finished, professional production. Color correction, special effects, audio transition smoothing—all these processes take place in the on-line edit bay.

Preparing your studio for scoring to picture, or adding an off-line room for "cuts-only" video editing, may be a fairly simple extension of your existing audio setup. Putting together an on-line room, however, represents an investment of time and money equal to, and probably larger than, whatever it took to piece together your bread-and-butter recording studio. Unless you're really ready to get into video in a major way, figure on providing an off-line system "in house," and get friendly with a facility where you can send your clients with their temp masters to make their on-line masters.

Turning Your Audio Studio Into An Audio/Video Production House. To get into video production, again you must make choices about how extensively you want to do it. Do you merely want to provide equipment for production, such as cameras, lights, or editing suites? Or do you want to write, produce, direct, edit, score, and mix the finished masterpiece?

The answer may be determined by how slick the end product must be. If you're the only person in town capable of making any sort of video whatsoever, and a band approaches you to make their video demo, you may get away with shooting it on a consumer camcorder and editing on a simple, consumer-market off-line system.

If, however, you're planning to make professional, broadcast-quality videos, get ready for something similar in complexity to mobilizing an army.

Today's standards for broadcast-quality music videos call for shooting on film rather than videotape (film still has a warm, "analog-ish" look that has yet to be matched in video). Shooting on film means buying or renting very expensive cameras and hiring well-paid people to run them. Film stock isn't cheap, and adequately lighting the shots to make your artistic statement requires professional lighting people and a good deal of time—and yet more money. As with audio, your final film image is several generations down from the original photography, so it's important to know when and when not to scrimp when getting the initial image "in the can."

Video can't be thrown together carelessly. To do it right requires talented people, time, and money—and a concept that's interesting enough to warrant the effort. Done right, however, a move into the world of video can be extremely lucrative, and creatively rewarding. •

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

> Considering A Move Into Video? Here Are Some Things To Consider.

BASIC TRACKS

that is just big enough to sit on the snare drum. Don't tape it down, just let it sit on top. This gets rid of any ring, and it works great!

Now that we've squared away the drum kit, next time we'll discuss different miking options. •

STUDIO CLINIC

stan cleaning.

First, when cleaning the capstan, do *not* put too much cleaning solution on the swab—excess solution must not drip down into the capstan motor. Most capstan motors have a rubber O-ring to prevent dirt and liquid from flowing directly into the motor bearings, but if a swab contains too much fluid, the liquid can flow over the O-ring and eat away at the capstan motor-bearing lubrication. Lack of capstan motor lubrication can cause early motor failure.

Second, not all capstans are made of stainless steel. Capstan motors on Sony/MCI machines and some Tascam decks use ceramic materials. Ceramic cap-

stans are great, in that they cannot become magnetized; however, they are a real bear to clean. For ceramic capstans, the best cleaner is Formula 409 household liquid cleaner, which seems to get oxide particles off places once thought impossible. This cleaner does leave some oil on the ceramic surface, so always use pure alcohol to finish off the job.

Lubricate your capstan motors regularly, according to the manufacturer's specifications. At least every six months, take the cap off the capstan motor and check for dirt, debris, and lack of lubrication. Several excellent brands of lubricant are available, but it's best to follow the manufacturer's suggestions. (This oil also is great for lubricating the jeweled bearings on the scrape flutter idler.) Be sure not to over-lubricate the motors.

Make checking the capstan motor one of your regular-priority preventive maintenance procedures. •

HANDS-ON PRODUCTION

Adding Overdubs. Congratulations! If you're ready to add more parts, your loop is playing perfectly in time—and that's half

Looping refers to the fact that the sample is retriggered immediately after it has finished playing, which gives the impression that it's playing in a continuous loop.

the battle. Now comes the really tough part: Getting the overdubs to groove with the human elements in the loop. There are no set ways to go about this, but plan on using a combination of quantizing, track shifting, and individual note editing. It's important to use a sequencer that offers high ppq resolution, and you'll find that advanced quantization parameters such as PERCENTAGE and SWING FACTOR will make your life a lot easier.

It's also helpful if your sequencer provides an edit selection filter, as this will allow you to quantize only those notes that fall within a certain number of clock pulses of specific beats.

Some sequencers also let you use a track as a template for quantizing other tracks. This can be a real boon, as you only

have to get one track right; all the other tracks can then be quantized to its values.

Don't forget that even though your loop may contain a full rhythm section, you'll undoubtedly want to have control over certain elements—kick and snare, for example—when it comes time to mix. To do that, you'll need to overdub these instruments, and once again, that means carefully placing them in the groove. The bottom line: Use a sequencer that provides all the proper tools, or you're liable to end up with the sloppiest groove this side of a teenage garage band.

Laying To Tape. MIDI is perfect, and MIDI timing is impeccable, right? Guess again. Depending on the density of data being played by your sequencer, you may find that your groove starts to slip as more parts are added. For this reason, you'll be better off laying only one sequencer track at a time to tape. To do this requires that you first stripe the tape with some sort of synchronization code. For my money, SMPTE is the only way to go. When you're laying a track to tape, mute all of the other sequencer tracks, or the MIDI data stream will be just as clogged as if you were trying to lay down all of the tracks simultaneously.

Occasionally you'll find that once the data stream is unclogged, a part that previously grooved perfectly now plays a bit too early. When this happens, simply shift the track back by the appropriate number of clock pulses (one to four should do the trick).

Wrapping Up. The art of looping comes from the streets, so the technological aspects haven't yet been adapted to the rigid digital perfection of the studio. As a result, if you plan on producing a hip hop tune, you're going to have to rely heavily on your ears to get you through. When it comes right down to it, though, worse things could happen. •

24 TRACK SHOWDOWN

TRACK

BY DR. RICHIE MOORE & BRENT HURTIG

WE PUT
THE OTARI
MX-80 AND
TASCAM
ATR-80
HEAD TO
HEAD

WELCOME TO THE TWILIGHT YEARS of analog 24-track recording. Standing its ground against a variety of incompatible digital multi-track formats, the professional standard of 24 tracks on 2" tape is very much alive and kicking—even if its prospects beyond this decade look dim. Yet digital's dominance over analog is inevitable. So why would anyone want to spend between $30,000 and $70,000 for a new analog 24-track? There are several com-pelling reasons. Currently, the only price-competitive digital alternative to analog 24-track is the Akai A-DAM DR1200, which we reviewed in the last issue of *EQ*. ($45,500 will get you 24 tracks on 8mm videocassette, in a system that works well but has yet to gain wide professional acceptance.) Otherwise you need to shell out around $100,000 or more for one of the digital 24-, 32-, or 48-tracks offered by Sony, Tascam, Studer, Otari, and Mitsubishi. At those prices, an analog purchase still makes sense for many commercial and well-equipped personal studios. →

Besides, analog 2" 24-track is a universal standard: Tapes can be interchanged freely among thousands of studios, and engineers with experience on one type of machine can quickly feel at home with another. And Dolby SR noise reduction offers analog performance that rivals digital quality. Then there are the many engineers and producers who prefer the sound of analog; recording drums, for instance, at a slightly too-high level on analog delivers a punchy, tape-compressed sound that can't be replicated with digital tape.

So, with the imminent arrival of digital dominance, a cost-conscious 24-track purchase may be just the ticket. Two machines that fit the bill are the Otari MX-80 and the Tascam ATR-80. But can 30 or 35 grand buy a fully professional machine? Has Tascam—a newcomer to the 2" 24-track field—built a recorder that can compete with one from Otari, a company with years of 24-track experience? *EQ* spent over 40 hours evaluating and testing the two, on the bench and in the studio at the Plant in Sausalito, California. We're ready to report the results.

OVERVIEW

YOU CAN CONSIDER OTARI'S MX-80 as a scaled-down version of their excellent 2" 24-track MTR-90II, or as a grown-up version of their 1" 16-track MX-70. The MX-80 entered the market in late 1987.

The Tascam ATR-80's design shows its lineage to the company's other professional machines, including the ATR-60 series and the DA-800 24-track DASH-format digital recorder (which was still in development when the ATR-80 was released in late 1988).

Both the Otari MX-80 and Tascam ATR-80 are available in 24- or 32-track analog format; the Otari machine also is available with an optional 2" 16-track headstack. Both are self-contained machines, with meters, audio electronics, and power supplies located below the tape transport. (Tascam also offers a meter bridge for "flying" the meters above the machine.)

The Tascam's audio inputs and outputs (I/Os) are switchable to operate at +4dBm or +10dBm levels: The latter is closer to the +8dBm levels run in many broadcast and video facilities, and allows for long cable runs with less chance of interference and noise. It's interesting that while the Otari also offers a second interface level, it's in fact a lower level: The MX-80 can switch between the "pro" +4dBm and "semi-pro" -10dBV levels (Otari evidently acknowledges that the MX-80 will find its way into some "-10" personal studios as well as traditional "+4" commercial studios). Both machines have actively balanced 3-pin XLR connectors (pin 3/hot). Unbalanced operation can be configured on the Tascam with a flip of a single switch. The Otari can run unbalanced, although jumpers must be moved at the outputs.

Physically, the machines are compact as 24-tracks go, although it takes four strong people to lift either of them (one person can roll them). The Tascam is larger, about the size of a commercial washing machine. The Otari, due to its smaller maximum-size reel (we'll get to that), is more the size of a household washer. Unlike a washer—which needs 240 volts—both the Otari and Tascam will run on 117 or 240 volts, with a 20-amp line. Neither manufacturer recommends a particular brand of laundry powder.

THE TRANSPORTS

CONSTANT TENSION? RELAX—BOTH RECORDERS employ capstan-driven *constant tape tension* transports. This approach ensures a constant amount of tension on the tape no matter how much tape is on either reel, which results in less tape wear-and-tear and a nice, even tape pack on both reels. Another added benefit of constant tape tension is that it's possible to rock the reels when the machine is stopped in EDIT mode, using a finger on the right-hand roller of the Otari, or the handy shuttle wheel on the Tascam. (If you're used to older 24-tracks, or inexpensive open-reel decks, it's practically magic to see one reel automatically take up the slack as you move the other reel in either direction!)

If you're a member of the video-game generation, you'll like Tascam's shuttle wheel: It makes it easy to "jog" tape and play it at variable slow speeds in either direction—great for locating specific passages. If you're from the old school and prefer to cue up a spot on the tape by grabbing the flange or hub of a reel with your hand, you'll find the Otari moves the tape more easily than the Tascam, which seems to exhibit more back tension. (This procedure isn't recommended by either manufacturer, however, since the reels are being moved against the wishes of the reel servos.)

I'm Your Capstan. Both machines use a brushless, Hall-effect DC motor to drive the capstan; in other words, a high-quality drive motor. [*Ed. Note: See "How It Works" in the Mar/Apr '90 EQ for a detailed look at a typical open-reel tape transport.*]

Unlike a Studer 24-track, which places the capstan shaft against the oxide (shiny) face of the tape, the Otari and Tascam locate the capstan against the backing of the tape, with the rubber pinch roller against the sensitive oxide. Most engineers favor this design, since a capstan tends to leave "skid marks" against the oxide when it engages—which could lead to program drop-outs, especially if the tape is started time and time again at the same spot (a common occurrence in video work). The trade-off of locating the pinch roller against the oxide is that it tends to pick up an amazing amount of oxide—particularly with Agfa tape, as our tests showed on both machines. Yet another reason to clean that tape path!

The MX-80 has a pro-oriented feature known as *pinch roller engage delay*. When you press PLAY or RECORD, the pinch roller doesn't press against the tape until the deck is up to play speed. The delay is only as long as an eye blink, but in the long term, it saves your precious master tapes from slight deformations as the pinch roller and capstan come together (with the tape sandwiched between them).

The ATR-80 has a different system called *soft start*. This delays the start of the capstan motor until the pinch roller is engaged against the tape. Since the capstan is what ultimately controls play speed, and Tascam's approach has the pinch roller and capstan meeting before the tape is locked at play speed, it's our gut feeling that the Otari design theoretically may cause less stress on the tape. We doubt, however, that either machine ever will cause excessive stress.

The Tascam's capstan shaft is ceramic, as opposed to the Otari's typical metal shaft. We like the choice of ceramic, since it doesn't need to be demagnetized, and seems to provide a better grip to the backing of the tape.

Of Speeds & Reels. So far, it may seem the ATR-80 and

MX-80 share a lot in common. On closer inspection, however, there are some important feature differences between the two decks. The Tascam ATR-80 is a 2-speed machine, with 15 and 30ips operation. The Otari MX-80 can be switched between a 15/30ips or 7.5/15ips "speed pair," from the rear of the unit.

The ATR-80 will hold 10.5", 12", or 14" reels. The largest ones provide up to an hour's worth of recording at 15ips; this is the professional standard for audio-for-video post production, where a single audio reel often must record an entire hour-long show. In this respect, the MX-80 is more limited, since it has a maximum reel capacity of 10.5", which will run for a half-hour at 15ips. (The 7.5ips speed *will* let the MX-80 run an hour, with slightly reduced performance—and lower tape costs.) For most musical applications, however, the preferred format is 10.5", 30ips or 15ips recording—no problem for either machine.

The reel motors of both machines really move tape quickly, as our fast-wind tests indicate, and generated relatively little heat. This pleased us, since heat is the prime culprit of many electronic failures. In additional to the ordinary fast-wind speeds, the Otari and Tascam both offer a *spooling* mode, which winds the tape at a speed somewhat faster than play. This mode packs the tape better than after a fast wind—almost as nicely as if it were played all the way through. It worked well on both decks.

The reel clampdowns that hold the tape against the reel platter are high-quality. They "self-center" the tape by expanding against the internal hub of the tape when turned (ATR-80) or when a lever is engaged (MX-80). A bit of reel play was noticeable with the Otari clamps, so we prefer those on the Tascam.

Thanks to their microprocessor-controlled transports, the two decks offer precision tape handling. In and out of any mode—PLAY, STOP, REWIND, FAST-FORWARD, RECORD, SPOOL (LIBRARY PACK on the MX-80), and CUE—they performed flawlessly and smoothly. Both machines illuminate the button of whatever mode is engaged. We appreciated the extra degree of visual feedback provided by the MX-80, with its two-level illumination: The lights are dimmed when "off," flashing when "ready" to perform a function (such as go into PLAY after a search), and lit brightly when "active." Unlike the buttons on the MX-80, those on the ATR-80 seem a bit too close to each other; it doesn't feel comfortable to operate them without looking at them.

Both machines will perform a SPOT ERASE—perfect for removing grunts and other unwanted sounds. The Otari also will play in reverse locked to speed: This is real handy when you have a piece of music that must end at a certain time, but don't know at what time it should fade in or start. (The Tascam will shuttle backwards, but not locked to speed.)

Both machines have a varispeed control (±50% on the Otari, ±20% on the Tascam), and, of course, can be controlled by an external transport synchronizer or another audio tape

How Much 24-Track Does $14,000 Buy?

TASCAM RECENTLY RELEASED the MSR-24, a 1" 24-track tape recorder with integral dbx noise reduction. We've been putting it through the paces in the studio and on the *EQ* test bench. We don't expect it to compete with machines costing more than twice as much (such as the Otari MX-80 and Tascam ATR-80)—and it's not designed to. But so far we're pleased with its performance and value. More on this revolutionary machine next issue. •

deck. Slaved via an Adams-Smith Zeta-3 synchronizer to a third 24-track and also a 3/4" video deck, both machines proved very responsive and quick to lock up, usually within a few seconds. Either machine would work well in an audio-for-video setting (don't forget the speed/reel size considerations discussed above).

Head-To-Head. As you would expect when spending this much money, the heads for the two machines are located on removable head blocks. A plug-in head block allows head replacement without having to perform time-consuming height, wrap, and zenith adjustments. Only the azimuth adjustment must be verified when changing heads, or optimizing the heads for a tape recorded on another machine. If you own a 32-track version of one of these machines, converting it to 24-track operation is a simple matter of unplugging the 32-track head block and replacing it with the 24-track head block.

The Tascam erase, record, and play heads (their order from left to right) are located really close to each other. Most machines include a *flutter roller* between the record and play heads, to minimize any vibration of the tape. The ATR-80 has its flutter roller located before the erase head; it's unusual to see it there, but its placement doesn't seem critical. The MX-80 heads are spaced farther apart, and the flutter roller is where you'd expect to see it. A nice feature on the MX-80 is two milled holes on the top of the head block cover to adjust the azimuth of the record and play heads.

Both decks sport a movable head shield—a shielded panel that lifts up from the deckplate to cover the heads and protect them from recording nearby electromagnetic interference (EMI) emitted by a fluorescent light or other EMI source. The Otari head shield is motorized, and moves up during PLAY, RECORD, or WIND modes, and drops in STOP. Slick! The Tascam head shield is engaged by hand, which isn't a problem, since once up, the shield needn't be retracted until the tape is unloaded.

Head crosstalk between channels on either machine is up to professional standards: Audio can be recorded right next to a time-code track without any problems. Heads on both decks are constructed of durable permalloy, and visibly were in perfect condition, without pitting or other flaws.

Under The Hood. Ideally, tape transports are set at the factory and never cause further worry. In the daily grind of studio use, however, transport serviceability is a concern, whether it's a capstan motor that needs replacement, or simply a fast wind speed control that needs a tweaking. How do our contenders rate?

The entire deckplate of the Otari MX-80 lifts up and can be held in place with a locking arm. The transport assembly is solid. Inside, the wiring is clean, and most major transport components are accessible and well identified. A nice job.

A look under the hood of the ATR-80 shows that Tascam has gone the extra mile to make a machine that will appeal to a service technician's aesthetics. The deck plate swings up ef-

fortlessly, with the aid of two pneumatic arms. Components are accessible, labeling is clear, wiring is immaculate and efficient, there's space for hands and elbows—all in all, a tech's dream. Tascam is even kind enough to include a test jig under the transport cover, for adjusting things like the solenoids and reel platter height. From a serviceability point of view, it may be the best transport interior ever built, in any 24-track, at any price.

Calibration of the Otari's transport is fairly straightforward, although not completely convenient. All the transport adjustment controls are on the rear of the transport, so you have to work from behind the machine. If the remote controls have been mounted somewhere else in the studio, it could be inconvenient to have to reach up and over the machine to change the transport status during adjustment—a process which took over three hours on the MX-80. Granted, the transport is factory-adjusted, and most studios don't even bother with further adjustments. But if the machine is moved from room to room on a regular basis, it's not unreasonable to optimize the transport every four to six months.

Like other Otari machines, the MX-80 has a series of tiny DIP switches that let the user customize the transport's setup, audio electronics, and remote functions. For example, on the transport board you can select the speed pair, or the way you enter and leave RECORD mode (using two buttons—PLAY and RECORD—or just RECORD). These switches require some time (and thought) to set up, but they could be useful.

The ATR-80's transport adjustment trimmers are thoughtfully located on a card accessible from the front of the machine. Full transport tweaking took under three hours, a bit less than with the Otari. As with the MX-80, there is an array of trim controls for speed adjustments, reel tensions, and so forth. A nice bonus on the Tascam machine is a test point and ground on the front of the transport card; it's not necessary, since an extender card could be used (as with the Otari), but it's a professional touch.

THE AUDIO SECTION

THE SONICS. As our *EQ* Lab Tests reveal, the Otari MX-80 and Tascam ATR-80 perform exceptionally well. A wide range of samples and music was recorded onto both machines. In all cases the two sounded just fine. Switching between INPUT and REPRO modes during RECORD mode (to compare the original signal with the just-recorded signal) was impressive:

EQ Lab Test

Product: Otari MX-80 24-track analog tape recorder.
Manufacturer: Otari Corp., 378 Vintage Park Drive, Foster City, CA 94494; (415) 341-5900.
Price: $29,950.00 including CB-140 remote. CB120 autolocator: $2,145.00. CB-120B time-code autolocator: $3,245.00.
Frequency Response (@ 0VU, 250nWb/m):
Claimed @ 30ips: 50Hz - 22kHz (±2dB); *@ 15ips:* 30Hz - 20kHz (±2dB).
Tested @ 30ips: 54Hz - 27kHz (±2dB); *@ 15ips:* 25Hz - 27kHz (±2dB).
Total Harmonic Distortion (THD @1kHz, 30ips, 320nWb/m):
Claimed: Less than 0.3%.
Tested: Less than 0.25%.
Signal-To-Noise Ratio (unweighted, @ 320nWb/m):
Claimed @ 30ips: 57dB; *@ 15 ips:* 55dB.
Tested @ 30ips: 60dB; *@ 15 ips:* 55dB.
Adjacent Channel Crosstalk (@ 1kHz, 0VU):
Claimed: More than 55dB.
Tested: More than 69dB.
Erasure Depth (@ 1kHz, +10VU):
Claimed: More than 70dB.
Tested: More than 78dB.
Wow & Flutter (peak DIN weighted @ 30 ips):
Claimed: Less than ± 0.04%.
Tested: Less than ± 0.01%.
Tape Speed Deviation:
Claimed: Within 0.07%.
Tested: Within 0.03%.
Fast Wind Speed (2,400' reel):
Claimed: 80 seconds.
Tested: 78 seconds.

What was coming off the tape sounded much like what was going onto tape.

We did hear an occasional slight difference between the two, though, in the form of a slightly accentuated high end off tape from the Otari; this could be attributed to the slightly greater amount of tape hiss found on the Otari.

The two machines employ similar "gapless" punch-in/punch-out circuitry that "ramps up" the bias circuitry gradually (over a matter of milliseconds) when you enter RECORD. If you remember the days when even the best tape recorders recorded a pronounced "click" whenever you punched in, you'll be thrilled to know that punches are virtually silent on both machines. You can even drop in on top of a note—the results may not be musically satisfying, but the punch will be clean.

Sonically, these are no-compromise machines, although the Otari MX-80 enjoys one advantage over the Tascam ATR-80: The MX-80 includes Dolby HX-Pro circuitry. *Not* a noise reduction system, HX-Pro is circuitry co-developed by Bang & Olufson and Dolby Labs to improve the headroom of magnetic tape. Although the HX circuitry can be switched out on the MX-80, we don't recommend doing so. Simply, the MX-80 sounds its best with HX engaged, particularly with a lot of high-frequency material and at 7.5ips. But we're splitting hairs: Both machines sound fine.

Calibrations. VU meters with peak LEDs are standard on both the Otari MX-80 and the Tascam ATR-80. These offer the best of both worlds: The mechanical VUs show an average level reading (excellent for calibrating the machine), while the peak LEDs illuminate to indicate peaks too fast for the VU meter to respond accurately. The Tascam's meters are a bit larger than the Otari's, and easier to read.

Audio calibration is necessary to ensure optimum performance from an analog tape recorder. It's essential to calibrate the machine whenever a new type of tape is used, and many studios calibrate at the beginning of every important session. Each audio channel has an associated *audio amplifier card* with a number of "trim" controls to set performance. Typical parameters include RECORD and REPRODUCE levels, RECORD BIAS (for optimizing output and minimizing distortion), HIGH- and LOW-SPEED EQ (to improve high-frequency response), and so on. Many of these controls can be set with a tone generator (found on most consoles), a plastic "tweaker" screwdriver, and the recorder's own VU meters. While most settings, such as BIAS and EQ, can be set for both high and low speeds on the Tascam and Otari decks, neither machine has individual level trimmers for both high- and low-speed SYNC, REPRO, and RECORD GAIN. In other words, some calibration is necessary whenever you switch between speeds.

Good machines, such as the MX-80 and ATR-80, provide

output level *test points* on each card, allowing the connection of an analyzer more accurate than a VU meter. Both machines calibrated very well using VU meters or an external meter. In this portion of our testing, we did discover some big differences between the two decks.

Let's start with the Otari MX-80. Its audio cards feel flimsy, without the metal reinforcement found on the Tascam ATR-80 and most other professional machines. And there are problems with the trimmers: It's hard to position the tweaker on the trimmers, and once you do, they feel sloppy and are hard to align. Fortunately, once aligned, they are indeed stable. These problems aren't a tragedy, and do not affect how well the machine records or its life span. They're simply a slight disappointment in terms of craftsmanship, and they make the service technician's job a bit more trying.

The Tascam ATR-80's audio cards tell a different story. First, they're among the most comprehensive audio cards found on any analog recorder. They include such bonus features as MIDDLE FREQUENCY COMPENSATION SWITCH, RECORD BIAS SIGNAL OUTPUT JACK, PEAK LED IGNITION POINT TRIMMER, and more. A total of 25 trimmers on one card! Quite impressive. The cards themselves feel solid, and the trimmers on the audio cards have a great feel, giving good tactile feedback, compared to the sloppy feel of the MX-80's trimmers. One area that could stand improvement: The BIAS trimmers and BIAS OSCILLATOR TUNING controls feel as if their screws are coming out during adjustment. They do adjust accurately, however.

REMOTES &
AUTOLOCATORS

EACH MACHINE COMES STANDARD WITH A REMOTE controller and has an optional autolocator. These components—the human interfaces that see constant use—give different machines a sense of character. The remotes for the Otari (model CB-140) and Tascam (RC-824) are similar. Their functions include: a duplicate of the transport controls; RECORD/SAFE selections for each channel; an ALL SAFE function; INDIVIDUAL or ALL CHANNEL STATUS modes (INPUT/SYNC/REPRO); a remote PITCH CONTROL with a TAPE SPEED readout; and more.

The Otari remote has a couple of extra features. First, it has four CHANNEL STATUS memories to store and recall different status selections for all channels at once. It also has a duplicate "mini-autolocator" function: The MX-80 has, on the transport itself, a 3-memory autolocator, complete with SEARCH and LOOP REPEAT functions—very handy for calibrating the audio cards. (The Tascam does have a RETURN TO ZERO and SEARCH TO CUE function on the deck.)

EQ Lab Test

Product: Tascam ATR-80 24-track analog recorder.
Manufacturer: Tascam, 7733 Telegraph Rd., Montebello, CA 90640; (213) 726-0303.
Price: $34,999.00 including RC-824 remote. AQ-80 autolocator: $2,150.00.
Frequency Response (@ 0VU, 320nWb/m):
Claimed @ 30ips: 45Hz - 25kHz (±2dB); @ *15ips:* 35Hz - 20kHz (±2dB).
Tested @ 30ips: 42Hz - 26kHz (±2dB); @ *15ips:* 37Hz - 27kHz (±2dB).
Total Harmonic Distortion (THD @1kHz, 30ips, 320nWb/m):
Claimed: Less than 0.5%.
Tested: Less than 0.31%.
Signal-To-Noise Ratio (unweighted, @ 1040nWb/m):
Claimed @ 30ips: 67dB; @ *15 ips:* 65dB.
Tested @ 30ips: 66dB; @ *15 ips:* 64.9dB.
Adjacent Channel Crosstalk (@ 1kHz, 0VU):
Claimed: More than 55dB.
Tested: More than 64dB.
Erasure Depth (@ 1kHz, +10VU):
Claimed: More than 75dB.
Tested: More than 78dB.
Wow & Flutter (peak DIN weighted @ 30 ips):
Claimed: Less than ± 0.05%.
Tested: Less than ± 0.03%.
Tape Speed Deviation:
Claimed: Within 0.05%.
Tested: Within 0.003%.
Fast Wind Time (2,400' reel):
Claimed: 100 seconds.
Tested: 106 seconds.

The Tascam AQ-80 autolocator is good, but nothing to write home about. In addition to standard features such as LOOP REPEAT and LOCATE TIME display, the AQ-80 has a STOP WATCH function, a PROGRAMMABLE PRE-ROLL (to allow for a pre-determined amount of lead-in time; handy for preparing a musician for an overdub), and 20 memories, programmable by keypad or on the fly from the tape counter reading.

The Otari CB-120 (or time code-based CB-120B) autolocator is sophisticated. It offers 99 user-programmable memories, a pair of programmable PUNCH-IN/PUNCH-OUT points (great for those tight punches!), PROGRAMMABLE PRE- and POST-ROLL, and more. This is a great autolocator.

CONCLUSIONS

THE OTARI MX-80 AND the Tascam ATR-80 were subjected to the same barrage of tests, technically and subjectively. They both held up extremely well under scrutiny, and both represent value for the money. So do we have a "winner"?

The Otari is a good machine, and for some studios would be the ideal choice. It feels substantial, sounds great, and handles tape extremely well. Its user interface—the remote and autolocator—is as good as any. Some people will want the MX-80 for the power of its autolocator alone; others will want it for the integral Dolby HX-Pro. It includes a number of wonderful touches: built-in minilocator, REVERSE PLAY, three speeds, and so forth. These features make the Otari an excellent performer—and it costs $5,000 less than the Tascam ATR-80.

However, the MX-80 lacks 14" reel capacity. It is a bit less serviceable than the Tascam ATR-80, and the trimmers and flimsiness of the audio cards were a disappointment. Overall, the Tascam is the better machine for most users. We wish it had a better autolocator, and the addition of Dolby HX-Pro and a 7.5ips tape speed option would be a *coup de grâce*. But these are quibbles.

For $5,000 more than the MX-80, you can get the better-built Tascam ATR-80, which has greater attention to detail. It's easier to work on. The 14" reel capacity makes it more "compatible" in the film and video worlds. In fact, anyone considering the purchase of an Otari MTR-90II ($39,950), Studer A827 ($47,500), or Sony APR-24 ($45,500) should consider seriously the more affordable Tascam. There are people who may still hold preconceptions about the sort of a 24-track Tascam can build. We call them "misconceptions." •

-20dBu—wide enough to handle any gear we connected to it. Individual Group and Master output modules are switchable internally for -10dBu or +4dBu operation.

Channels with microphone inputs employ a padless preamp with a single trim pot providing -2dBu to -70dBu of input sensitivity. This eliminates the more traditional resistive pad network, which may induce unwanted phase and harmonic distortion; we find the padless wide-range amp approach to be more effective when handling hot mike signals with wide dynamic ranges. Available on each mike input is +48VDC phantom power. All inputs are electronically balanced (but can accept unbalanced signals), with isolated transformer options for modules with XLR inputs.

Virtually every section incorporates an illuminated PFL (PRE-FADER LISTEN) switch that acts like a multiple solo function; these locking switches allow more than one audio element to be auditioned via the PFL bus simultaneously. Every section also has an illuminated ON switch (pre-aux).

Deluxe Input Modules. We expect the Deluxe to be the module of choice for most Delta owners. It has an XLR microphone input (pin 2/hot) and a 1/4" line input. Other front-end features include switches for a 100Hz high-pass filter and phase-reverse (handy for remedying phase problems caused by poor microphone placement, improperly mixed source material, or incompatible wiring).

The equalizer section provides 15dB cut and boost on four bands: fixed high frequency (12kHz) and low frequency (60Hz), with two sweepable mid-range pots (150Hz to

Soundcraft 200 Delta Console

SOUNDCRAFT'S FLEXIBLE NEW 200 DELTA CONSOLE is many things to many people. This compact, modular board features a number of improvements over its popular predecessor, the 200B. While it is equally adept at P.A., stage, broadcast, and recording applications, we will focus on its viability in the studio.

Mix & Match. The single-piece steel frame (rubberized on three sides) is available in four sizes—8, 16, 24, or 32 channels (with a rackmount option for the 8-channel version). Each frame includes a remote power supply, a master module, and a single ribbon cable with connectors for a complement of additional modules: The Group modules and up to four different types of input modules can be configured as desired. The only restriction is a maximum of four Group modules, installed immediately to the left of the Master module. The input modules currently available include Standard, Deluxe, Dual Line, and Stereo. (Delta modules are not backward-compatible with the 200B.)

We conducted tests on a 24-channel frame loaded with 17 Deluxe modules, one Standard module, two Stereo, four Dual Line, and four Group. It priced out at $8,400.

Common Features. Each module is self-contained and includes rear-panel connectors. This makes for easy maintenance and configuration (not to mention manufacturing). We were able to swap modules easily, by dealing with three screws and the locking ribbon connector on each module.

All the modules use Alps 100mm, long-throw faders and Alps pots, both of which feel solid. The board layout is extremely refined and overall construction is sturdy. Many switches incorporate LEDs, making operation much more intuitive.

The Delta works well with the mixture of -10dBu to +4dBu gear found in many studios today. All line inputs on input modules have a gain attenuator offering a range of +10dBu to

EQ Lab Test

Product: Soundcraft Delta.
Manufacturer: Soundcraft Electronics Ltd., Unit 2, Borehamwood Industrial Park, Rowley Lane, Borehamwood, Herts, England WD6 5PZ; (01) 27-5050.
U.S. Distributor: Soundcraft USA, 8500 Balboa Blvd., Northridge, CA 91329; (818) 893-8411.
Console Prices: $3,250.00 (8-channel w/ Standard modules) to $10,370.00 (32-channel w/ Deluxe modules).
Component Prices: Frames—$1,390.00 (8-channel); $1,970.00 (16-channel); $2,650.00 (24-channel); $3,230 (32-channel); Modules—$140 (Standard Input); $200 (Deluxe Input); $285.00 (Stereo Input); $225.00 (Dual Line Input); $185.00 (Group); $38.00 (Input blank panel).
Frequency Response (@+4dBu operating level): *Claimed:* 20Hz - 20kHz (+0, -0.5dB) (mike or line input to any output). *Tested:* Confirmed.
Noise (22Hz - 22kHz bandwidth, unweighted):

Claimed: -127.5dBu (mike input @ max. gain, EIN w/ 200Ω source). *Tested:* -125dBu (w/ 150Ω source). *Claimed:* -75 dBu (typical mix w/ 16 channels routed). *Tested:* -74.5dBu.
Crosstalk (@ 1kHz): *Claimed:* >90dB (channel muting). *Tested:* 86dB.
Total Harmonic Distortion (@+20dBu, 10Hz - 80kHz bandwidth, unweighted): *Claimed:* .005% (Deluxe input to mix out @ 1kHz). *Tested:* .004%.
Input & Output Levels: *Claimed:* +18dBu (mike input max. level). *Tested:* Confirmed. *Claimed:* +27dBu (line input max. level). *Tested:* Confirmed. *Claimed:* +21dBu (Group out max. level into 600Ω). *Tested:* Confirmed. *Claimed:* +-10dBV or +4dBu (Group out nominal levels). *Tested:* Confirmed.

2.4kHz, and 600Hz to 10kHz). A silent EQ switch patches the entire equalizer section in and out. This EQ section is reminiscent of the "classic" Soundcraft EQ found on such older consoles as the 2400 and 400B. It sounds great.

After the peak LED (indicating 4dB of post-EQ headroom) is a post-EQ effects loop implemented with a single stereo 1/4" connector handling both send and receive. (While more and more manufacturers are following this trend, it assumes the user has a patchbay that will accommodate a single stereo cable—otherwise the setup is a pain!)

The 200 Delta has a total of six auxiliary buses. Only four AUX pots are found on the modules: the AUX 3 and AUX 4 pots can be switched on the front panel of

each module to control AUX 5 and AUX 6. AUX 1 and AUX 2 can be jumped internally for pre-EQ, pre-fader, or post-fader; the rest are post-fader only.

After the fader come the pan pot and switches to route the signal to the Group 1-2 buses, Group 3-4 buses, and left/right stereo mix buses. Turning the pan pot hard left feeds Group 1, Group 3, and left stereo mix, while hard right feeds Group 2, Group 4, and right stereo mix. Finally, a 1/4" direct output is provided for sending a single channel's signal directly to tape or effect (bypassing the summing amplifier circuitry for a cleaner signal).

Standard Input Module. This module is a cut-down version of the Deluxe module. It lacks a direct output and switches for high-pass filter, phase reverse,

and EQ. Its 3-band EQ has only a single sweep-mid band (300Hz to 5kHz). Finally, AUX 1 and AUX 2 are permanently pre-fader— fine for some applications, but limited in recording. (Pre-fader sends are useful only for musicians' cue mixes: Post-fader control is desired for effects sends during mixdown or in a MIDI-based studio.)

Stereo Input Module. While the Stereo module has no microphone input, it does have two unique stereo input pairs (a total of four 1/4" jacks) with front-panel selection between the two pairs. This makes it ideal for use with today's stereo keyboards and effects; it also presents great potential for live and broadcast environments that integrate stereo playback devices such as CD players, tape decks, and cart machines (an optional internal RIAA phono preamp is also available). A MONO switch sums the selected left and right inputs. In this mode, the pan pot acts as a mono-in/stereo-out pan; otherwise it acts as a balance control between the left and right inputs.

A PHASE switch reverses the phase of the left input. The equalizer section is the same 3-band arrangement found on the Standard module, with the addition of an overall EQ switch.

Dual Line Input Module. This module crams two identical 1/4" line input channels vertically into the space of

one—ideal for tape and effects returns, or packing lots of inputs into a small space. The controls and signal path are necessarily simplified: there are no microphone inputs, effect loops, or peak LEDs. Output assignments are lacking, as well; signals go directly to the stereo mix bus. The controls are little more than GAIN, 3-band EQ, four configurable AUX pots, and VOLUME (a knob, not a fader).

Group Module. The flexibility offered by the input module selection is further enhanced by the versatility of the Group modules. This module has two separate 1/4" mono returns that can be used as tape monitors, effects returns, or additional inputs. Each has 2-band fixed EQ (60Hz and 8kHz), a single AUX 1 send, PAN, and VOLUME, as well as illuminated PFL and channel ON switches. There also is a 20-segment, wide-range LED meter (-24dBu to +16dBu) that follows the assignment of the module's lower section. The meter can be jumped internally for peak or average ballistics. (While average is what most people are used to, peak is handy when mastering digitally, since tape saturation is anything by desirable.) The fader feeds the GROUP OUTPUT (XLR), complete with a post-fader insert to add effects or compression to the mix. Typically, the first GROUP OUTPUT would be normalled to tracks 1 and 5, the second to tracks 2 and 6, and so forth.

The upper return always feeds its input directly into the stereo mix bus—*regardless of what the rest of the module is doing.* It could be used as an additional line input, but most likely would function as a tape monitor or for effects returns in the average session. The lower return has two additional

EQ Background Notes

Split-Monitor Console Design

The Soundcraft Delta is a good example of a common console design known as *split monitor*, or simply "split." The term refers to the location of the *tape monitor*—or *tape return*—section: The returns are located in an independent, separate section, usually split away to the right of the input channels, and almost always above the group output faders.

In a typical Delta console recording configuration, for example, there are eight tape monitors located above the four groups. TAPE RETURN 1 is located in the lower return section above GROUP OUTPUT 1(see review), and TAPE RETURN 5 is in the upper section above GROUP OUTPUT 1. RETURNS 2 and 6 are above GROUP 2; RETURNS 3 and 7 are above GROUP 3; and RETURNS 4 and 8 are above GROUP 4.

The returns are essentially line inputs, with a gain and pan control, and sometimes with associated EQ and aux sends (as with the Delta). They serve a critical function. During a typical recording session, the regular channel inputs are filled with microphones, synthesizers, samplers, and other instruments. That sometimes doesn't leave any channels open for tape tracks. The tape return section comes to the rescue, allowing you to monitor the inputs and outputs of the recorder without taking up precious input channels. In the traditional recording session, when it's time for mixdown, the tape deck outputs are unpatched from the tape monitor section, and take the place of mikes and instruments at the channel inputs. The returns then can be used for effects returns, if there are no input channels available for that purpose.

Now that synced MIDI instruments and affordable effects are on the scene, engineers need as many line inputs as they can get at mixdown. Any vacant tape returns not employed for bringing back effects are an excellent option—especially if they can be "flipped" to a long-throw fader, as four of the returns on the Delta can be. EQ and aux sends make them even more useful.

A popular alternative design is known as *in-line*; the tape monitors on this type of console are integrated into the same space as the input channels, so TAPE RETURN 1 is located in INPUT CHANNEL 1, RETURN 2 is in INPUT 2, and so on. (In many in-line boards, the group outputs are also integrated into the input channels; these are known as "I/O boards.") Because the returns on a split monitor console are in their own section, some engineers find them more convenient than in-line returns (even though this usually dictates a larger physical package).

—*Brent Hurtig*

switches—RETURN and SUB—that determine one of four functions.

With neither switch depressed, the lower section acts as a post-fader monitor for the GROUP OUTPUT; the LED meter reflects the GROUP OUTPUT level. Let's say you're recording a submix of drums coming in on multiple channels: With those channels routed only to a Group (channel MIX buttons not engaged), the Group fader can be set to the desired level to feed a tape deck, yet the drum submix can be monitored at a separate level via the lower return's volume pot (complete with EQ, panning, and AUX 1 send). Very convenient—otherwise, you'd have to compromise your recording and monitoring levels (or maybe you don't want to hear the drums at all).

The SUB switch turns the Group module into a subgroup; the fader and LED meter are tied to the amount of submix sent to the stereo mix bus (through the lower return's PAN pot) and the GROUP OUTPUT. This is the more traditional subgroup found on most consoles, and the least flexible of the four combinations.

The RET switch sets the lower section to act as a separate return fed directly to the stereo mix bus, just like the top module; the meter tracks the combined signal of the upper and lower returns, and the fader still controls the GROUP OUTPUT level. This gives you the ability to submix something like our drum mix to tape, most likely while employing the lower section as a tape return, through which the submix can be monitored.

Finally, with both RETURN and SUB switches depressed, the lower section is once again an independent return; the fader controls both the submix and the lower return, while the meter reflects activity in either return. This lets the lower section act as an additional line input or effects return *with EQ and fader control* during mixdown. This is cool!

Master Module. As you might expect, this final module contains the controls for the stereo mix bus, control room monitors, and master aux sends. Six pots are provided as master aux sends, each feeding one of the aux buses via a 1/4" jack.

The stereo mix bus feeds a pair of XLR outputs and a 1/4" mono output via the faders. A pair of stereo 1/4" jacks is provided for mix inserts—ideal for outboard compression of the entire mix. Each master aux send has an associated AFL (AFTER FADER LISTEN) switch; this is a "solo" switch, to monitor an individual send.

The MONITOR LEVEL pot feeds the post-fader stereo mix signal to a pair of 1/4" jacks (designed to feed a control room monitor amp) and a 1/4" stereo headphone jack. (Inserting a pair of phones "breaks" the output to the control room.) The 2-TRK switch provides for monitoring of an external mastering deck via 1/4" jacks. Any PFL or AFL signal will override either signal, should any corresponding button on the console be engaged. (This means you can "solo" something in the mix while monitoring through a 2-track. Nice.) A master PFL/AFL indicator lights to remind you of this monitoring status. The 20-segment LED meters (identical to those in the Group modules) reflect whichever of these three sources is being monitored.

A talkback mike is embedded in the panel, with volume control and switches for AUX 1-2 and ALL buses. This is good for talking to session players via foldback on the first two aux buses, or for slating tapes with comments. A variable-level, 1kHz oscillator tone is good news for Soundcraft 200B users who have

accidentally switched that console's fixed-level oscillator in at high volumes.

Conclusions. All in all, Delta's flexibility is its greatest asset; the four types of input modules can be combined as desired to accommodate specific needs. For example, 24 inputs could be placed in an 8-channel frame using Dual Line modules by sacrificing all Group modules. Dual Line modules and the returns on the Group modules can be used interchangeably for returning effects and up to eight tape channels (additional Dual Line modules could handle more, naturally). And their use as optional inputs on mixdown should not be overlooked.

There are a few negative aspects. The most unforgivable is a short-tethered remote power-supply that gives off an electric hum akin to that of an old fluorescent lighting fixture. [*Ed. Note: Soundcraft acknowledges this problem and claims to be rectifying it.*] On more subjective topics, there are no "remix" switches to route tape monitors to major input channels (although there seems to be provision for this on the circuit boards as a future option). We miss the ability to switch an attenuator into the PFL output, as we find ourselves constantly turning down the control room level to compensate whenever the PFL bus is engaged.

It would be nice to have external PRE/POST FADER switches for the aux sends, instead of jumpers. These would provide easy switching from pre-fader musician cue during tracking to post-fader effects (during mixdown). If this were not economically feasible, then we think it would be better to pair AUX 1/2 and AUX 3/4 on a switch, instead of AUX 3/4 and AUX 5/6; this would accomplish the same cue/effects switching.

Finally, the accompanying manual didn't offer much insight into applications for all the built-in versatility. A second, nominally priced, technical manual is required not only for schematics and the like, but for rather common information like module removal and jumper selection.

Specs-wise, we end on a positive note. Our *EQ* Lab Test not only purports excellent specs, but verifies most of them well within their tolerance (we did notice an output discrepancy of up to 1-1/2dB between similar modules). And while our *EQ* Lab Test was conducted at +4dBu, we put the Delta through its paces with a Tascam 80-8 8-track and 35-2 2-track (both are -10dBu) with no problems.

This console has a solid feel with a lot of nice touches—from illuminated switches and good controls right down to the plastic scribble strips, comfortable armrest and an appealing design.

The bottom line is that in addition to live and broadcast applications, Soundcraft's 200 Delta makes a great console for 8-track (and possibly 16-track) recording, live-to-DAT recording, or in a situation where additional submixed inputs are required for a larger high-performance console during a heavy session.

—*Jeff Burger*

Drawmer M500 Dynamics Processor

THE U.K.-BASED Drawmer company has a good reputation for building high-quality signal processors that do what they're supposed to do. Probably their best-known product is the DS-201 dual-channel gate, which is standard equipment in better recording studios.

Not a company to rest on its laurels (or Hardys), Drawmer has introduced a new breed of dynamic controller—the M500. While it's certainly not the cheapest dynamic processor available, this device performs extremely well, and may be a harbinger for other programmable dynamic controllers to come.

Overview. The Drawmer M500 is a dual-channel, multi-function, microprocessor-controlled, analog dynamic signal controller. (Memorize that sentence—there will be a quiz later.) Translation: The M500 is a two-channel unit with compression, limiting, expansion, gating, de-essing, and autopan and auto-fade functions. It can operate as two independent units, or both channels can be linked in a stereo configuration. Although the M500 is digitally controlled from its front panel or via MIDI, its signal processing is in the analog domain, as with traditional compressors and other dynamic controllers.

The single-rackspace M500 is well-built, externally and internally. The front panel contains an impressive assortment of numerous buttons, a large back-lit LCD display, a rotary data-entry wheel, and a four-button (up/down/left/right) cursor control keypad. The audio inputs and outputs are balanced, 3-pin XLR (pin 2/hot); the side-chain connections are 1/4" stereo jacks (ring = send, tip = return); the external trigger jacks are 1/4" unbalanced. Included are MIDI IN, OUT, and THRU jacks.

The M500 can be programmed via the front panel to operate at "professional" +4dBu or "semi-pro" -10dBu levels. A universal AC cord provides mains power; a back-panel switch configures the unit for 110- or 240-volt power.

Navigating The M500. At first glance, the giant LCD and array of pushbuttons may bring beads of sweat to the user's forehead. Since I hate to read manuals, I prefer knobs and self-explanatory front panels over the sort of multi-function "soft" keys found on the M500. But if there were a knob for each function available in the M500, the front panel would be the size of my whole rack! That being the case, I'm glad they did it this way.

After the initial shock of WYSINNWYW (What You See Is Not Necessarily What You Wanted) wore off, I found that the M500 is basically a very friendly machine. I would compare its data screens to the pull-down windows on a Macintosh.

EQ Lab Test

Product: Drawmer M500 Dynamics Processor
Manufacturer: Drawmer, Coleman St., Parkgate, Rotherham, S. Yorkshire, U.K. S62 6EL; (0709) 527574.
U.S. Distributor: Quest Marketing, Box 20, Auburndale, MA 02166; (617) 964-9466.
Frequency Response:
Claimed: 10Hz - 22kHz (+0, -1dB)
Tested: 20Hz - 22kHz (+0, -1dB);

10Hz - 30kHz (+0, -2dB)
Signal-To-Noise Ratio:
Claimed: 90 dB (20Hz to 22kHz)
Tested: 90.01dB (20Hz to 22kHz)
Total Harmonic Distortion:
Claimed: 0.08% (Worst Case)
Tested: 0.0005% (20Hz to 20kHz)
Headroom:
Claimed: +18dB
Tested: +22dB (over LEDs illuminate at +18dB)

When you press a button for one of the processing functions, information related to that function appears on the display.

For instance, when you press the function button marked COMPRESSOR, the first screen display shows high-resolution LCD renditions of input, output, and gain reduction level meters. To access parameters for the compressor, just press the "down" arrow on the front panel. The next screen displays the channel you are working on, various settings—including threshold, compression ratio, attack, hold, and release levels—and the type of compression algorithm being used. To change any parameter, use the left and right arrow buttons to position the cursor over the parameter you wish to change; a rotary soft knob located to the right of the display then lets you change the values with a quick spin.

This sort of programming approach applies to all the signal processing functions, as well as more global functions, such as MIDI utilities and overall operating levels. Once you have passed the initial familiarization phase, it's easy to cruise around through the displays and change everything. No worries, though: Recalling any factory preset will get you back into this universe and safe from the jaws of "-∞" attack times.

Thanks For The Memory. The internal memory consists of two separate blocks. The first block (memory locations 1 through 50) is the user block, where you may store your favorite settings for both channels of the M500. Anything that can be programmed from the front panel—including threshold settings for the gates, compressors and limiters, hold times, filter settings, patch order of processes, autopan limits, and autofade rates—can be stored and recalled. The second block of memory (locations 51-128) are factory presets. In fact, these may cover just about every possible use for the M500. The unit I tested came with a laminated sheet filled with descriptions of all the internal presets. The sheet looks much like the menu from the deli down the street, and may prove to be much more reliable. "I'll have the bacon and avocado with extra mayo and while you're at it, give me the VoxOvrDuck on Channel 2 of the M500."

Why A Duck, Why Not A Chicken? (Sorry, Groucho). VoxOvrDuck is a ducker setting suitable for voiceover work. Let's say you need to mix a radio spot for Larry's Used Shoe Store & Barbecue Pit, but you're having trouble with the music bed overpowering the voiceover. No sweat: Just patch both channels of the music through the M500, patch the vocal into KEY INPUT (external trigger), and select memory location 62. Now, when the vocal appears, both tracks of music will be "ducked" slightly— that is, they'll get quieter whenever the voice talks over the music. Hmm. . . maybe if I had enough of these M500s, I could get my records to mix themselves.

How about this one? You need a mix of your new demo that fades at 2:45 so it will fit between two cues in a video about toxic waste disposal. Piece o' cake. Run the whole mix through the M500 and select memory location 97. This particular preset will automatically fade your mix 2:45 after the M500 detects any audio above -10dB. Both times (from one second to 99 minutes) and trigger levels are programmable for this sort of effect.

Neat Tricks. The M500 lets you chain functions together in various orders. For instance, on one channel you can have the expander, then the limiter, then the de-esser, while on the other channel you connect the compressor feeding the limiter which feeds the gate. Any of these configurations, along with all the settings for each process, then can be stored in any one of the 50 user memory locations.

The gate process has a unique feature: It can record the envelope of any analog signal, then use that envelope to process any other sound. I recorded the envelope for a snare drum and then used the envelope to process a piano. It made the piano sound like tuned anvils.

The March/April '90 issue of *EQ* had a review of the Rane DC24 dual compressor/ limiter with a built-in crossover. The M500 contains four filters that can be configured to act as a crossover between the two channels of compression. I tried this and it worked flawlessly. Try this patch on a bass: Compressing the low end more than the high end adds a lot of fullness to the bottom end, without decreasing the dynamics of high notes or thumb slaps.

There are also two filter sections available that can be assigned to the gate, de-esser, or split-band compressor programs, for frequency-dependent processing.

MIDI Functions. The M500 supports MIDI, more so than just program changes from an external MIDI controller. The gate can be programmed to transmit MIDI note data when the gate opens, for instance, allowing external MIDI devices to be triggered. This lets you trigger a sampler from drums on tape. Gating or fading can be triggered from a range of MIDI notes. Pan ranges can be positioned by a MIDI keyboard. There's comprehensive SysEx implementation. The MIDI list goes on.

Sonic Performance. The signal path through the M500 is as short as it could possibly be. There is only one VCA in each channel. The digital processor implements the various gain reduction processes on the single VCA. The advantage of this configuration is that you maintain the shortest, cleanest signal path for your audio signal. Under traditional circumstances, you would run the signal through a gate, then patch it through a compressor, then a limiter, and the de-esser, and so on. Your award-winning sax recording is now passing through at least four VCAs and a gaggle of patch cords. I'd rather be attacked by a rabid gerbil than do this to anything I recorded. The disadvantage of using only one VCA is that cer-

tain routing configurations are not allowed. If you want the gate to be the first process in the chain, then you can't use the compressor or limiter—though hard gating would prevent any further processing. Oh well, it's always something.

Conclusions. The Drawmer M500 is a nice piece of gear, with a lot of functions crammed into a small space. With rack real estate becoming almost as valuable as property in Malibu, I welcome this direction in packaging/performance trends. I used all the functions on the M500, and they all performed as advertised. The gate sections were fast, the de-esser worked flawlessly (I had to use someone else's tape for that, 'cause I don't record esses), the compressors were smooth and unobtrusive, the limiter limited without obnoxious clipping, the fade faded, and the autopan made me throw up from vertigo.

I cleared out a space in my rack for an M500 and had the studio where I work buy three of them. Squeezing all those functions into one rackspace unit means that I will have room in my rack for more than one M500. By the time this review hits the streets, the Drawmer dealers' shelves should be re-stocked with M500s so you can check it out.

Now that I've finished this review, I am on a quest for the perfect power distribution system for my rack. My first choice would be cold fusion power with wireless distribution to all the components. I found something living in the nest of AC cords behind my rack, but I was told I couldn't kill it because it might be an endangered species. I could swear it looked a lot like a rabid gerbil.

—*Roger Nichols*

Stewart Electronics ADB-4 Direct Box

WHEN FIRST AS-SEMBLING the shopping list for a typical studio, one generally budgets only for the big-ticket items. Even when the microphone list is completed, direct boxes often are overlooked. Then, when they're needed, a mad dash is made to the nearest music store or pro audio supply house to buy one or two of these little boxes, with little or no consideration given to things like flexibility, performance, or even appearance. In time, the studio owner discovers that these boxes seem to be breeding, and five or six can be found in various corners of the studio, from the control room to the kitchen.

The people at Stewart Electronics are aware of this problem and have designed the neat, clean, and versatile ADB-4 in response. It's an active 4-channel device that solves the population explosion among direct boxes by integrating four otherwise separate boxes into one unit. →

EQ Background Notes

Almost all synthesizers and electronic keyboards can produce line-level signals, and can be plugged directly into most mixers. But what about other instruments that produce much lower levels? Some mixers, and all recorder/mixer "ministudios," allow you to plug in instruments directly, so you can record, for example, a bass guitar without having to mike its amplifier.

What if your mixer doesn't accept anything but mike and line levels? It's still possible to record an instrument directly by using a *direct box*, also known as a *DI* (for "direct injection" or "direct input"). A direct box takes an instrument's output (instrument or line level) and reduces it to a low-impedance, or *low-Z*, microphone level, which can be run hundreds of feet (from a stage or studio to a mixer) with minimal signal loss. It also balances the signal. The 3-pin XLR output of the direct box then can be fed directly into a balanced 3-pin XLR mike input. Most direct boxes are single-channel, though the Stewart ADB-4 is one of several multi-channel DIs on the market.

The average direct box has a 1/4" input jack and a 3-pin XLR output. Some, like the ADB-4, also have a 1/4" *foldback* or "output" jack, which is an unaltered output of whatever's plugged into the input jack. By connecting the output of this foldback jack into an amplifier, you can record an instrument directly while still hearing it through your instrument amp.

To deliver a balanced signal, direct boxes can be transformer-balanced (known as *passive boxes*) or active-balanced. Most engineers prefer the sound of active circuits, although a transformer-balanced direct box has two advantages: (1) since the transformer actually isolates the signal, it can prevent a nasty electrical shock from potentially faulty gear; and (2) passive DIs can run without batteries or external power. Just like condenser mikes, most active DIs are designed to run off an internal battery or phantom power.

Electric guitars and some other instruments get much of their distinctive sound from their amplifiers and speakers. Consequently, they often sound better if miked from the amp rather than through a direct box or regular input. However, for basses and even synthesizers, it's useful to mix a direct and miked signal together onto one track.

—*Brent Hurtig*

Anatek Pocket Products

These little boxes do much more than their names imply.

ANATEK MICRO-PRODUCTS' line of Pocket Products includes, at present count, ten MIDI-based "black boxes" that are smaller than a pack of cigarettes, weigh a few ounces, and start at $39.95. So they must be cheeseballs, right? Wrong. Closer inspection reveals that these mighty mites adequately perform their namesake functions, *and* some other functions that help solve common MIDI problems encountered in the studio.

Pocket Pedal. The Pocket Pedal ($109.99) was designed to expand the pedal capabilities of any MIDI device and to add foot control of MIDI parameters to those units without their own pedal inputs. A rocker-type pedal and a momentary-contact pedal can be used together, each one sending up to four types of control signals on any or all MIDI channels simultaneously. For example, a rocker-type pedal could simultaneously control MIDI VOLUME on Channels 1, 2, and 6, PITCH BEND on Channel 11, and MODULATION on Channels 8 and 14. At the same time, a footswitch could send SUSTAIN on Channels 2, 3, 5 and 10, PORTAMENTO ON/OFF on Channels 11, 12, 14, and 16, and a START or STOP command to your drum machine. This example may not describe a useful application, but it does illustrate the power available when slaying your own MIDI dragons.

Pocket Filter. The Pocket Filter ($109.99) provides filtering of just about any type of MIDI data you can think of. This comes in handy as a diagnostic tool. If you're having a MIDI data problem and can't pin-point the cause, use each of the Pocket Filter's filtering functions, one at a time, to isolate the type of data that is disturbing the receiving instrument.

Pocket Thru. One might assume the Pocket Thru ($49.99) is a basic MIDI thru box with one MIDI IN and three MIDI OUTS. But because it contains an *active* circuit, you can use two Pocket Thru units together for five MIDI OUTS. Or three Pocket Thrus for seven MIDI OUTS. Or—you get the idea!

Pocket Record. The Pocket Record ($149.99) is a 7,500-note, single-track MIDI sequencer that provides an audible FAST FORWARD function (the user can hear the track at high speed for cueing purposes). We discovered that it also can record note and SysEx data. This feature comes in handy when you want to do a quick bulk dump of your synth's program data so you can take it to another studio. Memory backup lasts for four hours, which should be adequate, assuming you don't get stuck in traffic. For those long drives, the optional Battery Plug accessory ($19.99) provides two years of backup.

Pocket Channel. This deceptively powerful MIDI channelizer ($109.99) was designed to perform two basic functions: filtering out the data on unwanted MIDI channels and remapping incoming data from one channel to another. The latter function, however, can be used to transmit data on multiple MIDI channels; incoming data on Channel 1, for example, could be set to transmit simultaneously on Channels 2, 4, 7, and 12.

The most obvious use for the Pocket Channel is to force an older MIDI synthesizer that transmits only on a fixed channel (or only in OMNI MODE) to transmit on one or more desired channels. Not so obvious is that you can use the device for MIDI channel switching on-the-fly during performance. A player can switch

---

EQ Lab Test

Product: ADB-4 Direct Box.
Manufacturer: Stewart Electronics, 11460 Sunrise Gold Circle, Suite B, Rancho Cordova, CA 95742; (916) 635-3011.
Price: $379.00 including AC adaptor.
Frequency Response:
Claimed: 20Hz - 20kHz (±0.5dB).
Tested: 20 Hz - 20kHz (-0.3dB, +0.2dB).
The results were well within the manufacturer's claimed response figures.
Total Harmonic Distortion @ 1kHz:
Claimed: 0.020%.
Tested @ +20dB: 0.056%; *@ +10dB*: 0.029%; *@ 0dB*: 0.041%; *@ 10dB*: 0.1%.
After our tests were completed, we

learned that Stewart Electronics tests the ADB-4 with a steep filter above 80kHz, to filter out inaudible frequencies. Without this filter, they confirmed our findings. In any event, we consider these amounts of distortion to be insignificant.
Noise (EIN):
Claimed: 100 dB, A-weighted.
Tested: Ground lift switch IN—98.3dB, A-weighted; Ground lift switch OUT—82.3dB, A-weighted.
Our test setup induced a ground loop so as to test the GND/LIFT function. As can be seen, it reduced noise significantly. Even the -82.3dB figure, however, is extremely quiet.

and 8-1/2" wide. If two of them are placed side by side, they can be installed in a rack with the front or rear panels facing out. Stewart offers the RMK-2 rackmount kit ($30.00) for this purpose, as well as the RMK-1 kit ($20.00) for rackmounting a single ADB-4.

Out of the box, the ADB-4 can be used as a free-standing, 4-channel direct box. Each of the four inputs is a 1/4" two-conductor unbalanced jack that accepts

signals from low-level instruments, such as guitars, and line-level devices such as electronic keyboards. In addition, a front-panel SENSITIVITY switch allows use of the speaker output of an amplifier. When the +20dB setting is selected, up to 1,000 watts from a guitar or bass amp can serve as an input. *Caution!* The ADB-4 does not act like a speaker load, so it's essential to attach the output of your amp to a speaker as well, to avoid damage to the amp.

Each channel has a front-panel input that is duplicated on the rear panel (labelled IN/OUT); they're wired in parallel, allowing an instrument to be plugged into whichever is more

Design & Construction. The ADB-4 is constructed of 3/32" metal, finished in textured black. It stands 1-11/16" high

between normal operation and the new channel(s) as required, manually or via velocity switching. Let's say Channel 1 is set for velocity switching. You could set the Pocket Channel to re-transmit on Channels 12 and 13 when the incoming notes' VELOCITY values are, say, 65 or higher, but re-transmit on Channels 10 and 11 when VELOCITY is 64 or lower. You could use this to switch between a normal and slap bass sound, bowed and pizzicato strings, lead and false-harmonic electric guitar, and a ton of similar effects.

Power Pack. All the Pocket Products obtain power from the MIDI signal when plugged into the master instrument (one reason for the small size and price tag). You do run into a problem powering any of these Anatek boxes, however, if the master's MIDI OUT port is not grounded—an oversight in some older MIDI units. The Anatek Power Pack ($39.99) provides MIDI power in situations where it is not available. Also, the Power Pack is useful when the total length of your MIDI cables exceeds 50 feet (the approximate point at which MIDI signals start to degrade). We found that the Power Pack can be used

alone to boost MIDI signals when 50 feet of spaghetti is already present. Add the device near the end of the line, and you're good for another 50-foot dash.

Conclusions. As you can see, there's a lot of power packed into these tiny boxes. We've only touched on six of the ten products in the family (Anatek also makes Pocket Transpose, Pocket Split, Pocket Merge, and Pocket Panic). And we just scratched the surface on solutions. After all, there are as many different MIDI setups (and accompanying problems) as there are users.

So before you go spending more than the cost of a good dinner for two on a MIDI widget, see if an Anatek Pocket Product will do the trick.

—David Bertovic

Pocket Products are manufactured by Anatek Microproducts, Inc., 400 Brooksbank Avenue, North Vancouver, BC, Canada V7J 1G9; (604) 980-6850.

powered that channel only. Therefore, if the user elects not to connect the PS-1, phantom power must be available on all the low-Z lines in use.

Performance. The ADB-4 performed well and quietly. The ground lift switch reduced noise considerably, as our tests show. The connectors are solid, and switches are flush-mounted to minimize the chance of damage. (These direct boxes do get kicked around!)

We were curious to see what would happen if the PS-1 were accidentally plugged into one of the audio IN/OUT jacks. No problems or damage were encountered, except, not surprisingly, the unit didn't operate. Simply plugging the PS-1 into its designated jack got the ADB-4 back on line.

Another question arises when the speaker output of an amplifier is used and the SENSITIVITY setting is not switched to +20. Thankfully, Stewart thought of this, too, and no damage to the ADB-4 will result. What you will get, however, is clipping galore, which can be corrected by simply moving the switch to the +20 position.

Conclusion. The ADB-4's advantages are many. It consolidates four direct boxes into one compact unit that can be rackmountable. It can be used as easily on the studio (or stage) floor as in the rack. Its versatility, especially in its ability to accommodate a very wide range of input levels with selectable output levels, means it can be used almost anywhere. A dedicated power supply plus phantom powering—freedom from batteries—adds to the list of desirable features. The ADB-4 is a fine value.

—David Bertovic

Intelligent Music Upbeat 2.0

BILLED AS "The Real-Time Rhythm Sequencer," UpBeat essentially offers the drum machine programmer a Macintosh-based visual editor that can be applied directly to any MIDI-compatible drum machine. UpBeat takes over the sequencing duties while firing internal sounds from your machine. Rev. 2.0 offers interesting additions to this useful program.

Computer-Aided Drumming. Like most drum machines, UpBeat works with patterns and songs. Performance events can be programmed into patterns by
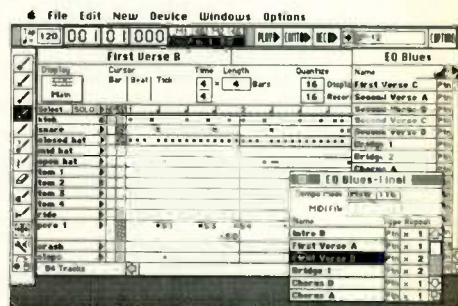
MIDI messages or by "painting" the attacks into a grid. After a series of patterns are created, they can be dragged into a song window for continuous play. In this respect, the program is like a "remote control" programmer.

UpBeat provides several powerful tools that can be called into play to alter, adjust, or enhance your creative output. Any individual event or group of events can be edited with respect to their position in the measure, attack velocity, and duration. In addition, events can be cut, copied, and pasted with a flexible variety of approaches.

If you think this is a neat idea, wait until you hear what happens when you start using

convenient. When a high-impedance instrument is plugged into one jack, the other jack can be used to connect it directly to the instrument input of an amplifier; the low-Z connector provides the low-impedance signal required for the mixer input. The GND/LIFT grounding switch on each channel affects both front and rear jacks, and can be used to reduce noise from the common, dreaded ground loop.

The rear panel provides one male 3-pin XLR connector per channel that outputs the low-impedance signal. There is a BALANCED LEVEL select switch with two settings: "-30" for connection to a mixer's microphone

input or "0" for direct connection to the input of most tape recorders.

Two 1/4" 24VAC power inputs/outputs, wired in parallel, are provided on the rear panel. These accept power from the PS-1 AC power pack included with the ADB-4. Either jack can be used when powering the unit, and up to four ADB-4s can be run off one PS-1, simply by daisy-chaining the ADB-4s with patch cords.

In addition, 48V DC phantom power from a mixer or power supply can take the place of the PS-1. Although not made clear in the Owner's Guide, we discovered that applying phantom to one of the XLR outputs

Legend
MIDI Note #
Note C D E
Standard – C0 (C-1)
Octave #
Alternate MIDI
Octave #

SOPRANO
CONTRALTO
TENOR
BARITONE
BASS

SINGING VOICE

PICCOLO
FLUTE
OBOE
ENGLISH HORN
B♭ CLARINET
BASS CLARINET
ALTO SAXOPHONE IN E♭
BASSOON
CONTRABASSOON
SOPRANO SAXOPHONE
TENOR SAXOPHONE
BARITONE SAXOPHONE

WOOD-WINDS

B♭ TRUMPET
FRENCH HORN IN F
TENOR TROMBONE
BASS TROMBONE IN F
TUBA

BRASS

SMALL
MEDIUM
LARGE

TIMPANI

HARP
GUITAR
VIOLIN
VIOLA
CELLO
BASS
BANJO
HARPSICHORD
PIANO

STRINGS

PIPE ORGAN

C0 (C-1) C1 (C0) C2 (C1) C3 (C2) C4 (C3) C5 (C4) C6 (C5) C7 (C6) C8 (C7) C9 (C8) C10
(Middle C)

20 40 60 100 250 500 1K 2K 5K 10K 16K
Frequency (Hz)

Home On The Range
Instrument Ranges, MIDI Notes, And Frequencies

STUDIO REFERENCE SERIES charts and diagrams are designed to be photocopied for handy reference in your studio work. This issue's chart centralizes many common reference needs—so you only have to look in one place to find the ranges of many traditional acoustic instruments, MIDI note numbers, and associated fundamental frequencies. Please note that while Middle *C* is always MIDI Note #60, different MIDI manufacturers refer to Middle *C* as *C*3 *or C*4. This chart uses *C*4 (black) as its primary reference for Middle *C*, in accordance with the U.S.A. Standards Association, and also provides the alternate reference of *C*3 (red) for the other half of the MIDI community.

—*Jeff Burger*

the "intelligent" goodies included in the program. For instance, UpBeat can generate its own random fills with the option of constraining or controlling parameters such as instrument, velocity, rhythm, and frequency.

New Features. The COUNTER window now contains two memory marker locations, a pre-count button, and a provision for tapping in a tempo. The TOOL palette contains new tools that let you paint in rests, loops, or even program change messages (very cool). The new audible WRENCH lets you hear single events while grabbing them for editing, and the MIDI-TO-MOUSE step-time recording tool lets you play MIDI events and place them graphically with a single mouse click.

Rev. 2 has significant enhancements over the original. The most important is the increase, within each pattern, from 32 monophonic tracks to 128 polyphonic tracks. This creates one *monster* rhythm computer, making it easy to arrange drum sounds spanning your studio's multiple instruments—including drum machines, synths, and samplers—into a single pattern. The maximum number of measures available in a pattern also has increased from 75 to 999.

Each UpBeat track is typically assigned a

Description: Computer-assisted drum scoring software for Macintosh.
Manufacturer: Intelligent Music, 116 North Lake Ave., Albany, NY, 12206; (518) 434-4110.
Price: $250.00 ($75.00 upgrade from Rev. 1).

MIDI channel and note number. Another new feature gives you the ability to overcome the 128 track/note limit by defining several MIDI note numbers to be available within a single track. This way, you might name a track something like "All D-110 Toms." Abbreviated indicators (S:1, S:2, S:3, etc.) show which sound corresponds to a given event. Each track will hold 128 different sounds. If there are more than 16,384 drum sounds in your life, see a therapist.

Without a doubt, the hippest new toy is the FOLLOW feature. When UpBeat creates fills on a track with FOLLOW enabled, the program "looks backward" up to eight events, and generates fills with notes that occurred in the recent past. Since UpBeat works equally well with synthesizers (firing pitches instead of drum sounds), this little trick can transform a fairly basic melody or bass line into something interesting and unique.

External Communications. UpBeat 2.0 is not an island. It can act as the master clock or slave to MIDI (MIDI Time Code isn't sup-

ported). It also has a very flexible MIDI File implementation. Patterns, songs, and performances with their accompanying variations can be exported to a MIDI File. While UpBeat can't read Format I MIDI Files (multiple channels, multiple tracks), the program does import Format 0 files (multiple channels, single track) into patterns. UpBeat had no trouble importing a 12-track MIDI File from Mark of the Unicorn's Performer; once imported and massaged, a full performance was captured and exported back into the original program. UpBeat is now compatible with Apple Computer's MIDI Manager.

Documentation. The manual is a pleasure and is thoroughly indexed. Even though some features are a bit tough to grasp at first, the tutorials and usage hints provide for a pleasant learning experience.

Conclusions. If you like drum machine programming, UpBeat will increase your enjoyment. If you hate to program, UpBeat may be your salvation. If you don't own a drum machine, UpBeat will transform your Mac and a multi-timbral sound module into one of the hippest machines on the block. When you consider all this program can do for the price, it's a good deal.

—*Norm Weinberg*

ADVERTISERS INDEX

COMING NEXT ISSUE

•USING DIGITAL TECHNOLOGY
•DIGITAL AUDIO WORKSTATIONS

DIGITAL

•HOW TO BUY A DAT MACHINE
•MIDI-ASSISTED MIXING

CLASSIFIEDS

INSTRUCTION

Help – I Can't Find My Masters!

REMEMBER WHEN THE MULTITRACK master tapes for one album couldn't all fit in the trunk of a car? Carrying four or five 2" analog reels in the trunk, plus the reel containing the alignment tones, would make the rear bumper drag over the railroad tracks on the way to the studio. And the *price* of a single roll of 2" tape would go a long way toward making a monthly car payment.

Media miniaturization to the rescue. Two-track masters of mega-buck recordings are being handed over to record companies on tapes not much bigger than the microcassettes used in dictating machines. I've heard that record companies are putting these DAT masters in 10-1/2" reel boxes or videocassette cases, just to keep from losing them.

The good news for your car's suspension is that multitrack recording is heading in the same direction. There are two new machines that record high-quality multitrack digital audio on tapes the size of 8mm videocassettes. If this trend continues, you won't even be able to *find* your multitrack masters in your trunk.

One of these machines, Akai's A-DAM DR1200 12-track digital recorder (reviewed in *EQ*'s Mar/Apr '90 issue), records 15.5 minutes of 48kHz audio on 8mm videocassette, and cannot (at this time) transfer digitally to other professional formats—it's sort of like a large multitrack DAT recorder. You can add up to two more transports for a total of 36 tracks.

Yamaha's new DMR8 8-track digital recorder will be out soon. It also uses 8mm tape, though the cassette is a proprietary design. I don't think that matters, as long as Yamaha keeps the supply flowing. After all, Sony is incompatible with Mitsubishi, which is incompatible with 3M; then there are the 2-track digital formats, all incompatible with each other. As long as you can go to a facility where you can copy it over to the format you want, you're fine.

Yamaha's 8mm tape holds 20 minutes of audio at the 48kHz sampling rate. Interestingly enough, the recorder is a stationary-head machine, as opposed to a rotary-head machine like the Akai, or a 2-track DAT machine. The DMR8 is actually a digital ministudio, consisting of a digital-domain console interfaced to the 8-track tape transport. The console has eight channel faders, which can control up to 24 tape tracks. (Up to two DRU8 8-track tape transports can be connected to the DMR8 for mixing down from 16- or 24-track.) Similar to the Yamaha DMP7 digital mixing processor, the channel faders are multifunction, controlling aux send levels, cue sends, and more. This represents a step toward the virtual console at a home studio budget.

Here's the part I have been saving for last: Yamaha's DMR8 is not a mere 16-bit machine. It uses 19-bit A/D and 18-bit D/A convertors with all internal audio represented as 24- or 32-bit values; 20-bit data is stored on the tape. Not bad, huh? If you're wondering what difference four bits can make, it's like the difference between the sound of a $150 CD player and one with a billion-times oversampling. It depends upon who's listening, but people's ears *are* getting used to better sound.

Dis · R · DAT. In an interview about two years ago, I said that rotary-head digital audio tape technology would make it possible for an artist to record a complete album at home and bring a stack of little DAT tapes to a studio, where they would be locked together and transferred to a digital multitrack machine for final mixing. This has come true. The duo Timbuk 3 did just that for their album, *Edge Of Allegiance*, which came out last September.

As they recorded each track on an analog machine, they simultaneously recorded on a Fostex D-20 DAT machine that was locked to the time code on the analog machine. The two transports were connected together, so when they would punch in on the analog deck, they would also punch in on the DAT machine. (The Fostex performs seamless crossfaded punch-ins because of its 4-head design.) When Timbuk 3 finished each track, the recorded information was on the analog tape for monitoring purposes and on the digital tape for later transfer to the digital multitrack.

After everything was recorded, Timbuk 3 took all the tapes to a commercial studio where, one at a time, the DAT tapes were locked up and transferred to digital multitrack for mixing. The only thing I don't like about the whole process is that I didn't get to do it first.

That's about it for now. I have to go out and help Walter Becker find the masters for the next Steely Dan album. He swears his girlfriend took them out of the trunk when she turned in the rental car. •

*Grammy award-winner **Roger Nichols** is chief engineer of Soundworks West Recording Studios in West Hollywood.*

> "Record companies are putting DAT masters in 10-1/2" reel boxes or videocassette cases to keep from losing them."

BY ROGER NICHOLS

ILLUSTRATION: SCOT HALPIN

FOR THE ELECTRONIC MUSICIAN WHO HAS EVERYTHING: THE WAY TO CONTROL IT.

ACTUAL DIMENSIONS: 19"W x 9¹¹/₁₆"D x 7"H

Actually, it's not even necessary that you have everything. Even if you have just a stereo keyboard, a sampler and a drum machine with separate voice outputs, you've already exhausted the capabilities of a conventional eight-channel mixer. And that's the reason behind the new M-160 and M-240 Line Mixers from Roland: modern electronic equipment demands modern mixers.
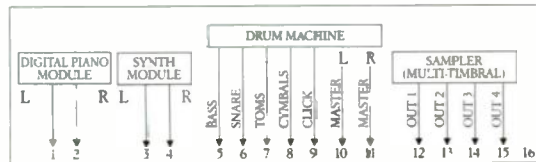
Never before have so many channels of control been produced in such convenient packages — perfect for the MIDI studio or the road. But it's the impeccable signal quality and low noise you'll appreciate once you put the M-160 or M-240 to work. The M-160 (16 channels) can be rack-mounted right in with your MIDI Modules and effects, while the sleek M-240 (24 channels) can be positioned in any electronic musician's set-up. M-160's and M-240's can even be ganged together for 32, 40 or 48 channels of control.

Each mixer has pro line level (+4 dBm) inputs and outputs, balanced XLR outputs and each yields exceptionally high S/N ratio, low distortion and outstanding frequency response. Each channel has input gain, peak indicators, panning control, smooth faders, three (count-em) stereo effect sends/returns plus 1 aux-send which can

WITH TODAY'S TECHNOLOGY, IT'S EASY TO FILL 16 CHANNELS WITH ONLY A FEW PIECES OF EQUIPMENT.

be positioned either pre or post master faders. There's also a Phone Mix In jack to facilitate headphone monitoring of the mix or a sequencer click, as well as master level meters, peak indicators, and mic level compatibility on channels one and two.

You might think with all of these features that we didn't leave a thing out. But we did — on purpose. There's no equalization. Why? Because MIDI keyboards and modules already offer much more tonal contour than the shelving equalization found on conventional mixers. Most importantly, adding EQ to a mixer inevitably adds noise — and the M-160 and M-240 were created with low noise as a design requirement, not an afterthought.

Nicest of all, you'll find the price tags on these mixers amazingly low for the features packed inside them. So, if your current mixer is starting to look like a wimp in the face of all your new MIDI equipment, it's time you checked out the new mixers that were designed specifically to handle the load. Today. At your Roland dealer. *RolandCorp US 7200 Dominion Circle, Los Angeles, CA 90040 (213) 685-5141.*

ACTUAL DIMENSIONS: 26"W x 11⅞"D x 3⅞"H

Roland
WE DESIGN THE FUTURE