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EQ

PREMIER ISSUE   MAR/APR 1990

41
COVER STORY

Now most anyone can afford digital-quality recording gear. That's great. But what happens when home studios decide to go commercial?

COVER ILLUSTRATION: ERIC LEE

Welcome To EQ

AS WE CAREEN FULL SPEED TOWARD THE 21ST CENTURY, there's a phrase I keep hearing more of, in conversations with friends, on the radio, in print: millennium fever. People use it to describe global, collective drives toward change that are intensifying the closer we approach the year 2000.

In your hands is a new magazine. I'm not sure it exists because of millennium fever. But I do know that it exists because of change.

As an engineer, producer, or musician, you're well aware of many significant changes that are occurring in recording technology, in techniques, in attitudes, and in creative expression. You're witnessing high-priced digital technology trickle down, and low-priced MIDI technology trickle up. You know that each year you're faced with buying new products, learning new techniques, furthering your career goals, mastering new technologies, and realizing your creative ambitions.

Now there is a resource you can trust to assist you with all of the change, learning, decision-making, and other challenges that you face. Welcome to *EQ*.

EQ is about creative recording. It speaks to those engineers, producers, and musicians who take their work in the studio *seriously*—whether that studio is an elaborate commercial facility, a smaller professional production room, or a personal studio. How can we speak to this broad a range of recordists? Simple: It's 1990, and a majority of these people work in a variety of recording environments. Furthermore, people committed to serious recording today share a vast number of interests in creative techniques, quality gear, career advice, and well-explained technical information.

Delivering information, however, isn't everything. That's why *EQ* accepts a compelling challenge—that our readers deserve editorial integrity in recording journalism.

Our integrity will be most obvious with our "In Review" section: All product reviews will be unbiased, candid, and free from any outside influence. We think you'll notice the difference in our reviews, starting with this first issue.

When you delve a bit deeper into *EQ*, you'll find that our commitment to editorial integrity runs from cover to cover. You'll see solid advice. You'll find interviews that are informative, and not just entertaining. You'll read investigative, in-the-trenches reporting that won't be afraid of offending anyone. You'll encounter strong opinions—some you'll agree with, and some you won't.

EQ doesn't exist in a vacuum. As a reader of this magazine your opinions matter immensely—and you're invited to participate in a dialog with us. Starting next issue, *EQ* will feature Letters To The Editor, as well as a Guest Editorialist. While we help you get better at what you do, you can help us do the same.

It's really quite exciting to be part of this new creative endeavor. I'm grateful to have such a high degree of enthusiasm and support from Phil Hood and the rest of the *EQ* staff, including those in administrative, production, advertising, and marketing roles.

Most of all, though, I'm grateful to have you as a reader. I think we're going to have a fine time chasing the tail of the millennium together.

Brent Hurtig

future watch

Digital Audio + MIDI + Computer Graphics = CD+MIDI

BY TED GREENWALD

One of the great unfulfilled promises of the digital revolution is interactive media. We *do* have the polling method used by tabloid TV programs to let the sofabound masses punch an 800-number into their telephones and sit in judgment of the latest front-page outlaw. But the integrated text, sound, and full-motion graphics of compact disc-interactive and digital video interactive have only barely left the lab.

On the horizon, however, if you shade your eyes against the blinding glare of MTV, you can see the hint of a medium in which the listener will be able to play an active role in the creative process—it's CD+MIDI, the brainchild of Warner New Media.

When played on a conventional compact disc player, CD+MIDI discs behave exactly like conventional CDs. Played back on a CD player equipped with a MIDI OUTPUT port, CD+MIDI offers 16-bit audio *and* a stream of MIDI data representing the music. To a limited extent, CD+MIDI also provides graphics for display on a television screen.

CD+MIDI audio channels are divided between vocals and backing tracks, so the listener can eliminate the instrumental music with a twist of his or her stereo amplifier's balance knob. Plug a MIDI synth into the CD player, and anyone's a producer: MIDI data can be routed to synths, samplers, and drum machines in order to re-orchestrate the music, or to a MIDI sequencer for a complete overhaul.

Turning Japanese

The CD+MIDI concept can be traced to the Japanese *karaoke* tradition, where bar patrons take turns singing songs ac-

EQ People

WITHOUT YOU, all of you, this magazine is but an inanimate construction of colored inks, flattened, clay-coated wood pulp, and glue. The real publication exists neither on the paper, nor in the ink that floats so lightly on the the surface of the pages, but in the hearts and minds of the readers: It is an ongoing conversation between them and the editors about the subject they share in common. Without that healthy, interactive, often passionate communication, there is no magazine. It takes two.

With that thought in mind I'd like to introduce you to some of the people who are responsible for this premier issue of *EQ*. If recording is your vocation, or your passion, you'll be having quite a dialog with them in coming months.

FEATURE WRITERS. Our cover story on the pro versus home studio controversy in Los Angeles is the work of **Cary Tennis,** an investigative reporter who lives in San Francisco. Tears For Fears was interviewed by **Andy Lyman,** a New York-based reporter on the staff of National Public Radio. **Michael Marans** and **Andy Widders-Ellis**, editors with our sister publication, *Keyboard*, penned the "Vocal Sampling" article, and the "Hidden Assets" features, respectively. Our "Near-Field Monitor Subjective Listening Test Of Doom" is the work of engineer and consultant **Rolf Hartley**.

Bryan Lanser, the product development engineer at Otari Corporation, wrote our "How It Works" feature.

COLUMNISTS AND REVIEWERS. "Future Watch" writer **Ted Greenwald** is a New York-based freelance composer, producer, and recording engineer. **Roger Nichols** is a three-time Grammy winner (Steely Dan and others) with recording, television,

Top row (left to right): Brent Hurtig, Roger Nichols, Richie Moore Ph.D, Jeff Burger. Middle row: Mike Collins, Norman Weinberg, Linda Jacobson, Phil Hood. Bottom row: Elisa Welch Mulvaney, Tom Lubin, Michael Marans, Steve Harlan.
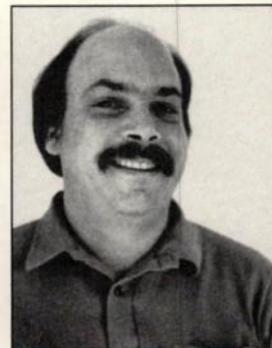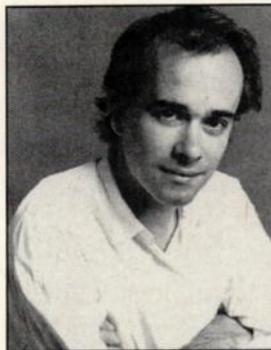
and film credits. He is chief recording engineer at Soundworks West in Los Angeles. **J. D. Sharp** is the respected owner and proprietor of Bananas At Large, a Northern California pro audio and music shop. **John Sugnet,** our "Self-Marketing" expert, is from San Francisco. He has served as consultant to numerous professional audio and music companies during the past ten years. Certified public accountant **Steven Harlan,** who writes for us from his home base in Portland, Oregon, also is a musician and personal studio owner. Percussion expert **Norman Weinberg** is an associate professor in the music department of University of Texas, Corpus Christi. He is a contributor to

Drums & Drumming magazine, and author of the book *The Electronic Drummer* (Modern Drummer Publications). Our "White Noise" columnist, **Freff**, has worked for the BBC as an American correspondent, done TV soundtracks and jingles, and is a graduate of the Ringling Brothers Barnum & Bailey Clown College, Class of '74. Freff is president of Uptown Technologies, a MIDI and audio start-up company based in rural Wisconsin. **Richie Moore, Ph.D.,** is chief technical guru at the Plant, in Sausalito, California. Richie's colorful background has taken him from sessions with the Beatles at Abbey Road to working with artists such as Huey Lewis & The News. **Tom Lubin,** a U.S. expatriate who writes for us from Sydney, Australia, teaches college-level courses in recording and video production. **Mike**
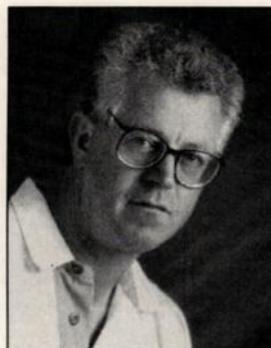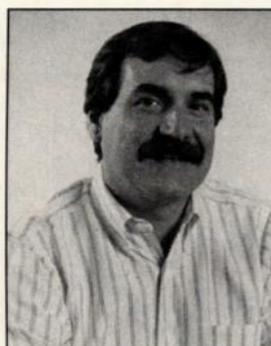
Collins works in London, where he is a freelance MIDI and recording engineer and contributor to Britain's *Studio Sound* magazine. Our audio/video section is headed by **Larry Blake**, a sound editor and re-recording mixer with Weddington Productions in North Hollywood. Among his recent projects are *sex, lies, and videotape* and *84 Charlie Mopic*. He is the author of the book *Film Sound Today*, the second edition will be published late this spring. Hollywood is home to **Rob Wait**, a post-production supervisor, film director, editor, and composer. Another filmland contributor is **Scott Gershin**, whose recent work includes *Honey, I Shrunk The Kids*. Reviewer **David Bertovic** is currently a L.A. session player and author.

STAFF. The basic *EQ* staff is headed by Editor **Brent Hurtig**, a Canadian transplant who is the author of *Multi-Track Recording For Musicians* (GPI/Alfred). Brent has worked with the recording industry in North America and overseas since 1977. Managing Editor **Linda Jacobson** is an ex-New Yorker who earned her degree in journalism at Boston University. Linda is former assistant editor of *Mix* magazine, and our resident hypermedia watcher at EQ. Senior Editor **Elisa Welch Mulvaney** hails from Boston. Formerly with *Frets* magazine, she's a multi-instrumentalist who rides the grammatical range at *EQ*, cutting stray typos and dangling participles from the prosaic herd. Our in-house bytes-and-digits honcho is Associate Editor **Jeff Burger**, a widely published author of articles, technical manuals, and books spanning computers, desktop media, synthesizers, and recording. Jeff formerly was computer consulting editor for *Recording Engineer/Producer*.

Additional assistance has come from my friend, *Keyboard* Editor Dominic Milano; and Advertising Director Dave Sweet, Publisher Charlie Baake, and General Manager Pat Cameron—whose support, inspiration, and faith have helped make *EQ* possible.

[signature: Phil Hood]

Research That Development

CD," WNM's Walt Klappert says, "the listener might miss it altogether. If you have time distortion when you're synchronizing audio tracks with MIDI, you have a problem." Nobody's talking about what *kind* of problem, or even who mastered the disc—the CD+MIDI mastering process is a closely guarded secret. Other manufacturers and mastering houses that want to get into it are on their own. Yet another challenge would lie in getting sound module manufacturers to standardize their voice/channel allocations so mastering engineers would know, for instance, that MIDI patch #7 would always be a piano-type sound.

Less ambitious than the audio are the graphics on the test disc. "Since so much of the bandwidth is taken up by the MIDI data itself," Klappert says, "only very restricted resources are available for graphics. Therefore the main use of graphics in CD+MIDI is to display lyrics, something you can almost always fit in." The song's lyrics appear one line at a time, in color, with a new line displayed every 12-1/2 seconds.

A graphic display of the music itself requires more bandwidth than CD+MIDI makes available. However, it *can* be produced by the listener, out of real time, by converting the MIDI data into standard notation via a suitable computer program. "We provide a score reduction—melody, chords, and bass—on MIDI channels 14, 15, and 16," says Denis Labrecque, Passport's VP of operations. "You can send the CD player's MIDI output to a computer running Encore [Passport's $595.00 music notation program] and see the music."

Shapes of Things to Come?

It's way too early to gauge CD+MIDI's impact. At the least, this format underscores the remarkable degree of acceptance MIDI has achieved, and the value of producers and engineers having a working knowledge of MIDI protocols.

Recording engineers who became conversant in MIDI while automating their signal processors may find a niche as producers of mass-market MIDI data, whether the medium is CD+MIDI, floppy disk, or ROM chip. Mastering houses will have to solve the problem of combining several real-time data formats onto a single medium. Artists will have to find interesting ways of using synchronous digital control of devices to enhance recorded music. Passport's Andy Muson predicts, "Things will spring from this—new educational applications, different kinds of entertainment. We're coming to a new turn in the road." •

T IS REMARKABLE that in the world of creative recording only a handful of products truly can be considered as "breakthroughs": devices that alter the methodology of production or redefine the sound of music. Equally amazing is the number of stillborn products that occupy no market niche and have no identifiable customer base. Granted, the process of advancing the state of the art usually is one of gradual steps, not massive leaps, but the number of me-too and meaningless products is staggering when compared with the number of innovative ones.

Several contributing factors make this so. The most obvious is that many manufacturers find it far easier to allow someone else to identify a need and demonstrate how to fill it, instead of finding it for themselves. For many firms, market research consists of going to trade shows, seeing what's new, and figuring out the most effective and rapid method to copy it. In some cases this isn't necessarily bad; the market for equalizers, for instance, is broad and deep enough to support products from many different sources.

But does every manufacturer of audio gear need to produce yet another 1/3rd-octave equalizer with exactly the same features as ten others? The sad truth is, it's far easier to construct some things (mixing consoles, for instance) than others (digital recorders). So that which is easy to build gets built, even if dozens of others are doing the same thing.

Another game played by manufacturers is to take a product and shuffle some features around to give it the appearance of a different device. All manufacturers are faced with the same basic equations—factoring the essential features and intended selling price—when it comes to determining the cost of producing a new item.

All too often the formula relies more on bells and whistles, and less on fundamental performance. Low-cost digital multi-effects processors are products of this type of thinking; for every good unit, there are two or three that could disappear without being missed.

Redundant products also spring from the perceived need of large manufacturers to be all things to all people. Management decides it wants to be the single-source supplier of everything from sound reinforcement mixers to studio monitors to power amps to tape recorders to synthesizers and beyond. It's nearly impossible to innovate in all these areas at once; instead we get a series of products designed to "compete head-on" with functionally identical products from other providers, with perhaps a sprinkle of originality in one or two items.

Irrelevant products seem to come about because of the "ivory tower" effect. A small group brainstorms about a product. Since these days the life cycles of some products are reduced to a matter of months, a premium is put on total secrecy so the competition won't get wind of the idea. It's like an underground guerilla cell: Knowledge of the product must be shared only with those immediately involved in the project. This paranoid method of operation without feedback from end users brings products to market that may miss their target entirely.

A synth that sounds great to everyone on the development team can fall flat on its face in the outside world. No matter how many years of experience may be shared by the team members—and how many degrees decorate their walls—the day they come to believe that input from working professionals is no longer necessary is likely the day they begin to engineer a product that will hit wide of the mark.

There are steps that manufacturers can take to improve their batting averages. The first is to accept the risk inherent in originality instead of lingering in the comfort of conformity. A balance can be struck between meeting existing needs with "safe" products (1/3rd-octave equalizers) and pursuing something

Manufacturers, take heed: Innovate, don't imitate.

BY J.D. SHARP

UP

DATE

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

Trident Vector

Who Cares About MIDI?

HIGH-PRICED digital technology is trickling down to the masses and low-priced MIDI technology is trickling up to the pros. We now have powerful and affordable samplers, MIDI-controlled multitrack recorders, and SMPTE-to-MIDI converters. But what about mixing consoles?

As early as 1985, some manufacturers (most notably Allen & Heath, Neotek, and Studiomaster) incorporated both digital routing and MIDI muting in some of their boards. These companies, however, were mavericks. Few builders at the low end of the market had the foresight to take advantage of microprocessor control; high-end console makers were largely snobs about MIDI, refusing to acknowledge its potential role in the professional studio.

Now, as we enter the '90s, a number of console manufacturers are finally delivering on the potential of both digital routing and MIDI control.

In recent years, **Trident** has been making a broad range of good-sounding, dependable consoles. Its massive, digitally controlled Di-An console recently re-established the company as a market innovator; the new **Vector 432** console ($110k to $135k list price) continues the trend.

For starters, the Vector's in-line monitor design has some nifty features, such as a parametric EQ section that can be split and routed to both input channel and tape monitor signals, as well as a BROADCAST mode that lets the short-throw faders (usually used as tape monitors) create a stereo mix independent of the multitrack sends.

But dig this: The 32-buss Vector has MIDI ports on its back panel. MIDI Time Code (or SMPTE) can automate the muting for the aux busses, the individual channels, the insert points, and more. The Machine

Control option will search tape machines *and* sequencers to bars and beats, directly from the console's computer. Very interesting.

From another British builder comes the **Soundcraft 6000** ($35k to $50k, including automation). This is the first of their consoles to offer an optional, integrated automation system that includes full MIDI implementation. Cue sheet-style lists can be set up to trigger MIDI events, and the board can follow MIDI or SMPTE time codes. And for a neat trick, the automation package includes a noise gate on each channel; when signals exceed or drop below the adjustable gate threshold, the system can be programmed to fire MIDI NOTE ON and NOTE OFF commands! Soundcraft may offer this system on some other boards, perhaps including the top-end **3200** ($72k to $91k).

At the other end of the scale, **Tascam** has created a pair of "ministudios" that offer some pretty powerful mute automation capabilities for very

little money. Both the **644** 4-track ($1,499.00) and **688** 8-track ($3,299.00) cassette recorders have microprocessor-controlled routing. Up to 99 "scenes" can be recalled by button or via MIDI; these scenes control channel muting, track assignment, input source selection, and more. The units also sport built-in MIDI-to-tape synchronizers.

Allen & Heath, whose CMC-series boards were among the first to offer MIDI muting, unveils two new consoles this spring. The first is the **Shadow** (expected to start at under $20k): Designed for 16- and 24-track recording, the in-line board will have MIDI-controlled muting, insert points, and EQ logic—and will also feature level automation as standard.

Also new from A&H is the **Vision**—a split-monitor console with MIDI muting as a standard feature. Prices for this board are expected to start at around $7,800.

Some manufacturers remain unimpressed with MIDI as a control option for larger boards. "Who cares about MIDI?" exclaimed one marketing director to *EQ*. "We automate our boards via SMPTE, and that's good enough." Perhaps he's right. Or perhaps he's right out of touch. •

Too Many Conventions?

INTERTEC, THE Kansas-based publisher of the trade publication *Recording Engineer/Producer*, organized a meeting of professional audio manufacturers from both sides of the Atlantic at the last Audio Engineering Society

Turtle Island Scores Small

THE TURTLE Island String Quartet (based in Northern California) recently finished recording the soundtrack for *Shock To The System,* an upcoming Corsair Films release that features Michael Caine and Elizabeth McGovern. Composer/synthesist Gary Chang had the "avant-jazz" group in mind right from the start, when he composed a minimalist score on his New England Digital Synclavier II using sampled strings, percussion, and gongs. Then, with charts to back them up, Turtle Island overdubbed acoustic string tracks *by ear,* while listening to the synthetic strings. The NED strings were then scratched from the final mix.

The 40-minute tracks were recorded in four days at Entourage Studios in North Hollywood. Engineer Bill Youdelman close-miked the instruments using AKG C-12s and Neumanns running through preamps straight into the back of a Mitsubishi digital 32-track recorder. The musicians used special headphones, which had only one earpiece each, to bring more "feel" to their performances.

Turtle Island founder Darol Anger describes the musical style of the black comedy's soundtrack as a "pizzicato atonal tango with Italian restaurant violin on top." Took the words right out of our mouth. •

The Turtle Island String Quartet: David Balakrishnan, Darol Anger, former member Laurie Moore, and Mark Summer (current member Katrina Wreede played the session).

Convention (in New York) to discuss a common viewpoint—there are too many trade shows in this industry. Key issues include exorbitant costs, overlap with related industries, not enough personnel, schedule overlaps, and inappropriate locations.

A typical trade show year is a bevy of acronyms: Winter NAMM in Anaheim, CA; Summer NAMM in Chicago or Atlanta; AES in N.Y. or L.A. and Europe; Musik Messe in Germany; NAB and CES in Las Vegas; SMPTE in L.A. or N.Y.; MIAC in Toronto; and more.

According to a spokesperson, the group's goals are "to constructively reorganize the exhibition calendar globally, to consolidate shows to strengthen the appeal of the necessary exhibitions, to improve the quality of attendees, and widen the base the shows appeal to."

The organization's first move is to establish which major shows its members will attend or boycott. As this movement gains power and momentum, you may witness the end of those conventions that don't adapt to the industry's needs.

• • •

In other trade magazine news, the 1989 Technical Excellence & Creativity (TEC) Awards ceremony, sponsored by California-based *Mix* magazine, raised a record-setting $34,000. The proceeds benefited the House Ear Institute of L.A., Audio Engineering Society Educational Foundation, and Full Sail Center for the Recording Arts. Hats off to our friends at *Mix.* •

New Microphone Technologies

AUDIX HAS unveiled the MIDI-One dynamic microphone for use with digital samplers. The $399.00 mike puts out a +4dBm line-level signal—good news for anybody concerned with clean sampling. (Many samplers don't offer mike-level inputs, forcing the user to go through an additional boosting stage such as a mixer or tape deck. Those that *do* have mike inputs often still have cleaner circuitry associated with the line inputs.) For some, the choice of dynamic design may be a plus, offering a warmer, more natural sound than condensers.

The MIDI-One must be phantom-powered—a unique feature in dynamic mikes. It works with any *external* (not console-integrated) 9V to 52V power supply. This transformerless design helps meet the manufacturer's goal of eliminating any "middle men" between the mike and sampler.

Audix claims the resultant specs are 40Hz-20kHz with 120dB

Audix
MIDI-One
sampling
mike

signal-to-noise ratio and 5 volts output level! We don't recommend singing in the shower with this one.

With the advent of digital samplers and portable DAT players, stereo miking is more popular than ever. AMS is shipping a new stereo mike, the ST250, which can be customized to suit a variety of needs, including X-Y or M-S format, vertical or end-fire mode, variable effective capsule angle (0-180 degrees), and variable polar pattern (omni through all cardioids to figure-8). Other controls include attenuation, bass roll-off and stereo invert.

All controls are situated in a separate control unit, allowing for remote adjustments after mike placement. The control unit also houses a headphone monitor and batteries for 40 hours of self-powering (mains or phantom power can be used also). We can't wait to test the manufacturer's claims of linear 20Hz-20kHz frequency response on and off axis, and find out if it really does have "the most accurate stereo image of any microphone available." We'll keep you posted.

Puzzled by the terms "X-Y" and "M-S"? You'll find more information on stereo miking in our review of the Crown SASS-P stereo mike. (Our review section starts on page 66.)•

Music Software Matures

MUSIC SOFTWARE has come a long way in a short time. One of the most obvious examples is the Darwinian progress of editor/librarian software used to alter and archive programs within MIDI devices, via MIDI's System Exclusive (SysEx) bulk dump standard.

Several software manufacturers have started to overcome the "old" problem of needing a different editor/librarian program for each MIDI device that contains programs which a user wants to archive or edit. **Opcode** has incorporated their large stable of SysEx programs for individual instruments into a single generic librarian called **Galaxy** for the Apple **Macintosh.** Any existing or future modules for specific MIDI devices (70 modules to date) can be accessed from the one main program. Opcode has an excellent track record for ongoing support; however,

a programming language is incorporated to allow users to write their own modules for addressing virtually any MIDI device with SysEx implementation.

Hybrid Arts takes the concept one step further with **GenEdit** for both the Macintosh and Atari ST, adding editing features to the generic librarian framework. As with Galaxy, device-specific SysEx drivers are supplied by the manufacturer or written by users via a built-in programming language. This package also has a sort of generic patch editor "construction set"—including graphic representations of sliders, knobs, switches, and graphs with which users can build their own remote front panels for MIDI gear.

Both programs provide the ability to send PROGRAM CHANGE commands to MIDI switchers, automating the proper connections for MIDI handshaking. This eliminates the problem of constantly plugging the appropriate cable into the computer's MIDI IN, given the typical multiple-instrument setup. GenEdit brings this idea a step forward by taking an "electronic snapshot" of an entire studio of MIDI settings at

a step forward by taking an "electronic snapshot" of an entire studio of MIDI settings at the press of a button; this file can be used later to instantly restore the gear to those same settings.

What's the bottom line? We predict that while Opcode has hands-down archiving support for virtually any instrument worth its volume control, GenEdit will give Galaxy a run for its money when editing is the main concern.

More good news: The MIDI File Standard is definitely alive and well. This protocol for importing and exporting MIDI sequences between different programs on different computers has appeared in almost every recent sequencer offering. **Passport**'s **MasterTracks** has been ported from the Mac to the Amiga, IBM-PC, and Atari, all with good ol' MIDI File Standard. The **Opcode** release of **EZ-Vision**, a simplified version of their popular Vision sequencer, offers similar support (as does the parent program).

Although the existence of the MIDI File Standard has pretty much meant that songs created on one sequencer can be played and edited on another, at least one other application exists. **Encore** from **Passport** imports any sequence file adhering to the protocol *and* displays it as standard music notation. It does it pretty well, too! It's a pleasure to see a founding principal of MIDI—standardization—maintain its role as technology evolves. •

Evolving Recording Media

WHILE 2" tape is certainly not a thing of the past, a plethora of new recording media is giving it a run for the money. For example, **Tascam**'s new **MSR-24** puts 24 tracks on one-inch tape for $13,999.00. The machine takes 10-1/2-inch reels, operates at 7-1/2 or 15 ips, and incorporates dbx Type 1 noise reduction. Worried about crosstalk? We've heard it, and it's amazingly quiet.

The MSR-24 is the latest result of Tascam's quest for more tracks on smaller tape. Their greatest achievement in that area has been squeezing eight tracks onto a standard high-bias cassette in the **688 MIDIStudio** and **238**.

Toa also offers an 8-track cassette format with the **MR8-T**. While neither of these represent the ultimate in audio quality, they provide good sound and features previously unavailable for the price.

Digital is coming down in price, with an associated deluge of tape formats. The soon-to-be-released **Yamaha DMR8X** uses a proprietary 8mm video cartridge in a format that combines an 8-track digital recorder with a 24-channel automated console, for around $45,000. The design incorporates a stationary head, which has fewer moving parts than the rotating head design found in most videotape-format machines such as **Akai's A-DAM** (reviewed on page 66). Even if delivered with the most impressive list of features and specs, the proprietary tape format undoubtedly will cause

Yamaha
DMR8 digital
multitrack

potential buyers to reflect on possible tape shortages and format life.

Akai, on the other hand, has seemingly learned from past experience, employing standard 8mm tape cartridges for its A-DAM digital 12-track recorder. The big news from Akai, however, is the release of the **DD1000**—the first stereo digital recorder on the market based on Sony's 650MB rewriteable optical disk technology. Approximately 20 minutes per side can be recorded digitally at 44.1kHz or 48kHz sampling rates. External drives can be accessed for recording and playback via SCSI, and a proprietary cable is used to exchange data between up to four DD1000s for functions such as digital transfer and back-up. Randomly accessed samples can be triggered manually or via an internal SMPTE cue list. The unit supports simultaneous stereo playback—two additional tracks can be recorded while the first two are playing back. Its retail price is expected to be under $15,000. Look for more about this much-talked-about advance in the field of disk-based recording in future issues.

As if there were any doubt left that digital disk recording is the way of the future, manufac-

turers of traditional tape recording products are entering this arena at an increasing pace. For openers, **Studer Editech** was formed last August when Studer, the well-established tape machine manufacturer, acquired Integrated Media Systems and the Dyaxis digital hard disk recorder. The company has already announced new products, including the **Dyaxis 2 + 2**, which can employ a second audio processor to play back and/or overdub a total of four digital tracks simultaneously.

Otari has revealed its commitment to codevelop a digital disk recorder with **Digidesign**, which is known for Soundtools, the disk-based recording system. We'll let you know the details as the product nears the market.

While these advancements certainly are exciting in terms of their varied price points and user levels, we do offer a caveat regarding standardization. Few of these advances in the price/performance ratio use established media; when they do, they rarely employ a standardized format. Hopefully, the industry will forget neither the benefits of standardization in its analog past, nor the problems created by the format wars in the video world. •

Tears For Fears

Sowing The Seeds Of Self-Produced Success

BY ANDY LYMAN

THE HURTING became "the waiting" for fans of Tears For Fears. The duo's last album, *Songs From The Big Chair*, sold eight million copies and yielded two Number One singles in the United States. Then four years passed before the group produced *The Seeds Of Love*, released last autumn on PolyGram.

Like many English bands that started out in the late '70s and early '80s, the Tears—Roland Orzabal and Curt Smith—sported shaved heads and a heavily synthesized sound. They were barely distinguishable from their contemporaries. *The Hurting* and the single "Shout" from *Songs From The Big Chair* earned them the description of gloom rockers. *Songs'* second single, "Everybody Wants To Rule The World" was bouncier. It locked in their reputation.

There followed the inevitable months of touring the States. While in Kansas City, Roland and Curt reportedly experienced a musical revelation when they heard a woman named Oleta Adams singing in a hotel bar. As they put it, their souls were sanctified, and so were planted the seeds of "Seeds Of Love."

After the tour the Tears returned to England and took a much needed hiatus. They also tried to get *The Seeds Of Love* off the ground, including a prolonged stint in the studio with Chris Hughes, producer of *The Hurting* and *Songs From The Big Chair*. But the sessions didn't click. They parted with Hughes and decided to produce the recording themselves, with the help of stalwart engineer David Bascombe.

Casting away their old dependence on synthesizers and samplers, Tears hired a cadre of great session musicians, including Phil Collins and Robbie MacIntosh. With the inspirational Oleta Adams on piano, drummer Manu Katche, and bassist Pino Pallidino, they laid down basic tracks live in the studio, which is located in Roland's house. Among the results are two contrasting and stirring duets between Roland and Oleta Adams: "Badman's Song," with its Lowell George-style phrasing, and "Woman In Chains." The title cut, "Sowing The Seeds Of Love," is a Beatles homage/tribute *à la* Sgt. Pepper's and *Magical Mystery Tour*. "Advice For The Young At Heart" incorporates the upbeat attraction of both The Style Council and Prefab Sprout. "Standing On The Corner Of The Third World" is ambient, and features a solo horn by Jon Hassell. "Swords And Knives," though rejected by director Alex Cox for the *Sid And Nancy* soundtrack, echoes as the perfect coda to that movie. "Year Of The Knife" is colder than a winter day in England, and "Famous Last Words" is an uplifting conclusion.

At the end of 1989, with Christmas less than a week away, Tears had spent a month recording "Do They Know It's Christmas II" and rehearsing the difficult new material for their current tour. America had just invaded Panama—to Mrs. Thatcher's applause—and Romania's head had yet to fall. Curt had just moved into a new house and was staying at a friend's while carpenters had their way with power saws in his living room. Roland had spent the afternoon

TEARS FOR FEARS

Christmas shopping. It was just another day for "sowing the seeds of love."

• • •

EQ: *The talk around town about the making of this record concerns how you finally found your soul for* The Seeds Of Love, *especially since you had tried on two separate occasions to record, using your previous methods, and you shelved those attempts.*

Curt Smith: We didn't finish anything. We attempted a few weeks with one lot of producers, which never worked out because we weren't on the same wavelength. Then we went back again to working with Chris Hughes. That went on for a long time, for a multitude of reasons, one being that we'd always worked with Chris and felt comfortable with him, so maybe we were blinded to the fact that it wasn't working out this time around. But after about eight months, we came to a point when me and Roland listened to the tracks and didn't really like what we heard. We'd been using the same old methods of sitting around the Fairlight every day and putting everything in order, and everything was clean, everything was nice. It worked on *Songs From The Big Chair* because we were into it, but this time we weren't. We found it all a bit sterile, a bit

too programmed, and all too nice and neat.

I think the songs demanded some more guts and attention, which meant taking it to a different level than just sitting around a computer and meticulously putting together backing tracks. So we decided to start again. We felt it was time to start producing ourselves and try out different methods of working.

EQ: *Was there trepidation involved in producing the album yourselves with engineer David Bascombe?*

Curt: No, we didn't feel that. We felt it was the most natural step to take. People say, "Wasn't it a hard decision?" And I always say, "No, it was the easiest decision to make." Because the way to make it hard for yourself is to work under conditions you don't want, and try to do something you don't want to do, because you're always fighting against yourself.

EQ: *At the same time, did you feel pressure about the time lag between albums? People were saying "Four years! My god, when are they going to have another album out?"*

> *"If we'd released this album, the way it sounds, a year after Songs From The Big Chair, it would have confused the hell out of everyone."*
>
> **—Curt Smith**

Curt: No, I never really thought about it too much while we were doing it. I just wanted to get something done that we enjoyed. The success of the last one afforded us the luxury of not having to worry too much. If we'd released this album, the way it sounds, a year after *Songs From The Big Chair*, it would have confused the hell out of everyone. We've been four years away and you understand why when you listen to it: "Okay, they've done a bunch of changing, a bunch of growing up." We haven't spent the whole time sitting on our hands. Plus we needed the four years off. We wanted a bit of normality in our lives for a while.

EQ: *To record the new album, did you set up and play live in the studio?*

Curt: Well, myself and Roland mainly watched a lot of it. We got together a bunch of musicians that we'd wanted to work with for a while. There were different recording sessions because we used a quite diverse amount of musicians. The first sessions were with Manu Katche, Oleta, Carole Steele playing percussion, Pino Palladino,

and a guy called Simon Clark playing keyboards.

Basically we just played them the songs, then handed it over to them and let them express themselves as musicians. Then we'd pick out bits we liked or guide them in the right direction. So it was a lot of experimenting, 'cause we didn't want to limit ourselves to one way of working and limit the people to playing exactly what we want. We spent a long time just sitting around, going through everything until we felt comfortable with what we were hearing.

EQ: *Nevertheless, did you go back and use any sequencers or any MIDI stuff?*
Curt: We did, yeah, for something like "Standing On The Corner Of The Third World," which is just a little drum machine and then the band kicks in. Also on "Woman In Chains," we had a nice groove going on a little drum machine. As long as it felt right, that was the only criteria. With those two it felt right to start off with a little groove going on a drum machine.

There's a drum machine on "Third World" and the Fairlight on "Woman In Chains." We introduce the band later [in those songs] so in that sense, the band actually played on top of what we had. Phil Collins is on "Woman In Chains" and Manu Katche comes in on "Third World," though he does play a bit of percussive cymbals and things over the beginning.

All of it was done pretty much on the Fairlight. We had that big Yamaha DX, a DX1 originally, then we got a rack for it. As long as it sounds good, that's what matters.
EQ: *Where was it all done?*
Curt: We recorded a lot of this at Roland's house, 'cause we just put a Trident desk in there, and we had a 32-track digital Mitsubishi X-850 downstairs in his hallway.
Roland Orzabal: Basically my house is what's called a "studio house." It was origi-

nally a small art school. At the top of the house was the artist's studio, which is a room with double-height windows so it's a nice and light working space. In there I have a Trident Series 80 desk. It's 36-channel, 24 monitoring. During the album we were sending tielines down to the hallway where we had the Mitsubishi 32-track digital, and sometimes two Mitsu's, so you could barely walk in the hallway. I think 32 tracks is a good number, and actually we were working 64-track. We couldn't have fit [the album] on 48 tracks. That's basically the only reason for two 32-tracks, although I have worked with the Sony digital 24-track machines and they sounded a lot more digital than the Mitsubishi.
EQ: *Does the Mitsubishi machine have particular features you really like?*
Roland: We did a lot of live recording and we would bounce between machines; you can edit drum tracks till you're blue in the face. You don't have to lose anything, because you've done a digital copy and you're not splicing tape. You can repeat two bars of a good fill throughout a couple of verses—that kind of thing. Bouncing and editing between the two Mitsu's was sort of key to a lot of the backing tracks, like on

Devon Bernardoni

Doobies Reunion Album Gets This Engineer's Career On Track

BY BRENT HURTIG
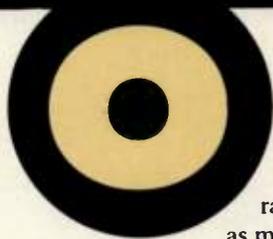
N THE SURFACE, hers seems like a story of big luck. A young woman moves from Vancouver to San Francisco to train race horses. She likes music as much as she likes horses. A few years later, the former trainer finds herself in the saddle as co-engineer for the reunion album of one of the 1970s' most popular groups. Suddenly her work is climbing up the charts; the music she has recorded is heard on one of the top music videos of 1989.

At 28, Devon Bernardoni has reason to be happy. The gig that has secured her engineering career is *Cycles* by the Doobie Brothers. Released in the autumn of 1989, the album is now certified gold, and its sound is classic Doobies: solid drums, thick electric and silky acoustic guitars, polished and up-front horns, and crystal-clear vocal harmonies. But her chance to record such trademark tracks didn't come easy: As every good trainer knows, it takes commitment, hard work—and yes, a bit of luck—to succeed.

"I started by recording graveyard sessions. I went to school during the day, and I had to sup-

port myself at the same time with an evening job. That first year was the worst time in my life," says Bernardoni, leaning against the large-frame Trident TSM console in Studio B of the Plant, in Sausalito, California. "But I knew when I decided to be an engineer that I was going to follow through with it. Eventually I was able to get enough studio work to support myself."

A pianist and drummer, Bernardoni first strayed from the stable to the studio in 1982, when she applied to be an assistant at Starlight Sound, a 24-track studio located in Richmond, California. She graduated a year later from the College of Recording Arts, and shortly thereafter moved on to Different Fur Studios in San Francisco.

The next few years offered a wealth of training opportunities with a diverse range of talented artists: the late technodance king/queen Sylvester, the acoustic Clubfoot Orchestra, rock 'n' roll straight-aheads Cris Loiter & The Hangouts, and everything else from jazz to children's music.

An important break came in 1987, when Bernardoni received a call from producer/engineer Jim Gaines—best known for his work with Huey Lewis & The News, Santana, Starship, and Journey—asking if she'd be available as an assistant. It wasn't long before she began to share engineering responsibilities with him.

When Gaines was called to engineer the Doobies, he brought Bernardoni with him. But he left early on in the project to move to Nashville, leaving her to co-engineer the sessions with three other studio veterans: producer Charlie Midnight (Joe Cocker) and songwriter Eddie Schwartz, and later Rodney Mills (38 Special). Bernardoni says, "I was the only one, besides the members of the band, to start and

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

ACTUAL DIMENSIONS: 19"W x 9¹³/₁₆"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.
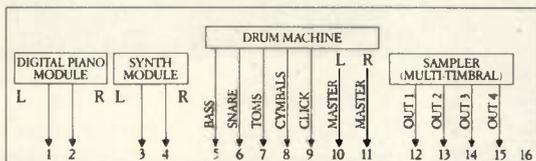
ACTUAL DIMENSIONS: 26"W x 11⁵/₁₆"D x 3⁷/₁₆"H

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WE DESIGN THE FUTURE

Duane Hitchings

Songwriter/ Producer Navigates The Charts In His Floating Studio

BY JEFF BURGER

WE'VE HEARD of people keeping their music careers afloat, but songwriter and producer Duane Hitchings has taken the concept literally—his personal studio is located on a boat. Many of the songs he's penned with Rod Stewart, Steve Perry, Eddie Money, and a host of others have been spawned on "Troubadour," his 42-foot Roughwater cabin cruiser anchored in Marina Del Rey, California. It's not unusual for Hitchings to weigh anchor and happily bob about a few miles offshore or cruise to nearby Catalina Island while fishing for new tunes.

Maintaining a studio on a boat requires some considerations that are unknown to the shore-bound. First and foremost, electrical de-humidifiers are a necessity to fend off moisture. For those sea-going times when Hitchings can't access the dock's AC power, he uses a spare 18-horsepower diesel engine to run a generator that pumps out eight kilowatts of power for studio use. Every single piece of gear is strapped down. The rear stateroom houses the studio; while it does suffer from propellor vibration, it is rela-

tively stable. "If anybody else is crazy enough to do this," Hitchings says, "don't put your gear in the front of the boat, because that's where the real rock 'n' roll happens!"

The space restrictions of a floating studio are certainly a consideration for someone who plays guitar *and* a large stack of MIDI keyboards. A 12-track Akai MG-1212, with its combination mixing board/tape transport, helps to save space. Monitoring is handled by JBL 4311s for recording and by shelf-mounted Yamaha NS10Ms that he uses for mixing.

Vocals on Hitchings' demos are all recorded with Neumann TLM-170 condenser microphones in the same studio/control room. "It's funny when a singer stands up and then freaks when suddenly there's a current shift! All in all, I was worried about how my clients would feel about coming on the boat, because this is how I make my living. But Steve Perry defined it when he said, 'You won't have trouble getting me on the boat, but you'll have trouble getting me off!' It's fun, it's new, it's something special, and there's a lot of fresh air down here. People love it."

The MIDI portion of the studio centers on Hitchings' newest toy, a Simmons SDX that he uses for sequencing, sampling, and drum sounds. "I love the user interface on the SDX—the trackball, two buttons, and menu-driven screen. Most other manufacturers are not building their screens big enough, and a lot of these low-end 'workstations' can't sample, either. Then again, the larger workstations have too many features and are just too expensive. Stereo sampling on the SDX would be nice, but stereo is not all it's built up to be. My bass is not stereo; drums are balanced and run across my board. I drive a lot of stereo outboard gear instead. It's a whole lot easier and cheaper to go out and buy five or six rack-mounted synths for $1,000 apiece than it is to constantly hassle with stereo sam-

pling."

On or off shore, Hitchings' credo is experimentation. "If you really want to make your tapes sound good, experiment with your equipment and find some way of setting it up that nobody else will. I'm coming to the conclusion that there's no such thing as instruments—there are sounds. There's things that go boom, things that go sizzle, and then there's all the other pretty stuff. And let's mix all these sounds together."

When it comes to production, Hitchings draws upon his years of experience playing with Alice Cooper, Rod Stewart, and the Buddy Miles Express, when he learned "what a producer could do to destroy an album! A producer should never try to take over the entire project and put his will on something that isnot there in the first place. The producer's job is not to amplify and let the artists—no matter how good or bad they are—express themselves, and give them the framework. The producer is like a coach who encourages the talent and makes all the doors open up."

As Hitchings sets sail to reel in Joan Jett, Greg Kihn, Hall & Oates, and Whitney Houston with his newest batch of hooks, he casts out a final tip for any landlubbers who might be interested in acquiring studio sea legs. "If you like to fish, be sure to crank the bass up, because the vibrations through the bottom of the boat attract schools of fish. There will be one sure market for your songs—in bait and tackle shops!" •

IT'S ABOUT TIME

Real Drummers Don't Beat Buttons

BY NORM WEINBERG

WHAT EXACTLY IS MUSIC? Easy question, tough answer; we could say that music is a series of sounds organized over a period of time. You can't have music without sounds and you can't have sounds without taking up time. This column will be devoted to those aspects of the music-making process that relate to sequencers, arpeggiators, time and clock converters, and other time-based devices. Today's timely topic is drum machines.

• • •

Timing and music. It seems that the pendulum of change has started another swing. When drum machines were first introduced, the notion of having a recorded drum track with "perfect time" was manna from the gods. Perfect was hot. Recently, manufacturers have been trying their best to bring back "human feel." Now perfect is cold. Machine-generated rhythm tracks sound pedestrian, lifeless, and just plain dull to many ears. But this doesn't have to be the case.

Beat It. To program like a drummer, you've got to think and act like one. First off, "real" drummers don't play little plastic buttons with their fingers, they use sticks. The most realistic way to program a drum machine is from a drum pad, period! Even if you've never held a pair of sticks in your life, the purely physical sensation of hitting something is going to impart a certain amount of realism into your programming. (It's great therapy, too.) So, beg, borrow, or steal (well, maybe not steal) some sort of percussion controller immediately.
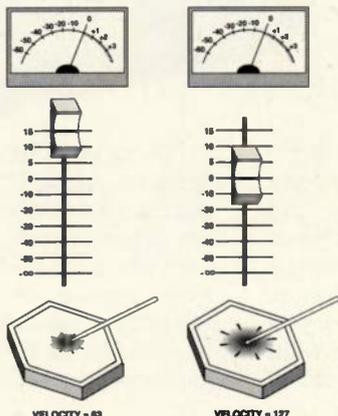
The Long & Winding Pattern. Drum machine manuals suggest you conserve time and memory by working in short patterns of one to two measures that can be called up in various sections of the song. But drummers don't think in small chunks. Every bar flows into the next, and like snowflakes, no two measures are exactly the same. Try setting your machine's pattern length to eight or 16 measures, or even longer if it makes sense.

Air O' Dynamics. Don't hear quite enough dynamic variation? Try adjusting the pad controller to a different velocity curve or level setting. When programming from pads, your loudest stroke should have a MIDI VELOCITY under 127 (the maximum velocity setting). Drummers are always capable of hitting a drum or a cymbal just a little harder for those special situations where the music needs an additional kick in the butt. If you're peaking the velocity at anything under strokes that could kill mortals, adjust the velocity curve.

By the same token, playing and recording at too soft a velocity level could result in a higher noise floor, since you might be required to crank up the fader of the drum machine track to get a good mix among the other instruments when you lay the final version to tape. The trick then becomes striking a balance between a good dynamic range and good overall level.

Let Me Hear Your Body Clock. Programming a great "feel" into a machine is actually a matter of phrasing. We haven't yet addressed the problem of quantization (auto-correcting your performance to the closest specified note value, e.g. eighths or sixteenths). Here are some other experiments you can try:

First, set the metronome to a faster click if possible, such as sixteenth-notes (change the resolution, not the tempo). This provides a more intricate framework in which to program feeling. Second, do all your drum machine programming in high-resolution mode by turning the quantization off; after all, drummers don't have quantizers! You can capture the subtle dynamic nuances by programming from pads, but you need to be in high-resolution mode to capture any sensitive timing variations.

Try the previous suggestions with the hi-hat first. If the result is close to what you want, bring the machine into step programming mode and fix any fluctuations in the tempo that seem to get in the way. Once you're happy with the result, play in the bass drum, the snare, and any other sounds you want in your pattern, using the same techniques. As before, go back into step mode to fix any offending attack points. Don't fix them all—just the ones that bother you.

The end result of all this is a pattern that sounds more like a drummer than a machine, with variations in dynamic contrasts and timing resulting from musical expression. These are just a few of the tricks and techniques that can be applied to drum programming. Next month, we'll cover a few creative techniques that can be applied to computer-based sequencers. Until then, keep organizing those sounds over time! •

VELOCITY = 63   VELOCITY = 127

Higher MIDI velocities yield cleaner mixes.

> "Machine-generated rhythm tracks sound pedestrian, lifeless, and just plain dull to many ears. But this doesn't have to be the case."

ILLUSTRATION: RICK EBERLY

WHITE NOISE

We Are All Beginners

BY FREFF

YOU DON'T KNOW WHAT you're doing. I don't know what I'm doing. *Nobody* knows what they're doing. Not in life, and not in love, and certainly not in the recording studio.

Assuming I now have your attention—and I should, after that frontal assault on your ego—let me explain. Charlie Chaplin's last great film, *Limelight*, has a scene where Chaplin and Buster Keaton play aging vaudevillians who are busy making themselves up in a backstage dressing room. Chaplin's character is quiet and focused. Keaton's character complains about everything, especially the small size of the dressing room. What do they think we are, he bitches—*beginners?*

To which Chaplin says, offhandedly, "We are all beginners. None of us lives long enough to become anything else."

Yes!

Chaplin was right. Even restricting ourselves to the single topic of recording leaves us with more to explore and learn than can fit into a lifetime.

But that's not depressing. It's liberating. It leads directly to some encouraging conclusions and useful techniques, among them:

• **What you know is more important than how much you know.** We've all heard recordings that were technically perfect but emotionally bankrupt, or carbon copies of somebody else's hit. Maybe we've even made a few. But there's no excuse for it. Knowing how to operate your gear is far less important than knowing how to operate it with taste and originality.

Don't be content to swallow a technical manual, or the Gospel According to Anybody, then spit it back whole during your next session. Question everything. Accept nothing. That response chart that came with your hot new microphone? Challenge it! Maybe you'll *like* the way the mike sounds when jammed into the upper left-hand corner of the room. How will you know unless you try it? And that drum fill you adore on the radio? Avoid it like the plague. How will you ever find an original voice if you only say things that have been said before?

Tackle this: For one whole day in the studio, try only the unfamiliar. Invert all your tried-and-true reverb settings; close your eyes and mangle the EQ on one or more channels, then work with what you get; restrict yourself to incredibly cheesy effects boxes, or mikes you've never used.

In short: Do nothing that you actually know how to do. If you surprise yourself, even once, you win.

• **What somebody else knows, you can learn.** If the Special Theory of Relativity can be made clear with the right diagrams—and it can—then you have no worthy excuse to avoid anything in Audioland. Not even stuff buried in jargon (like Cross-Interleave Reed-Solomon encoding of compact discs) or apparent Zen mystery (like what the heck "ground" is).

Complexity is often simplicity in a fancy suit—or else plain simple-mindedness, like the sampler operation manual that starts out with "Step 1: Reinitialize the operating system defaults." (English translation: turn it on.) When you hit such snags, translate them into real-world terms, then step through the actions for yourself, or draw diagrams, or (if you're taking a class or working as an intern) brazenly demand that your teacher run through it again from a different angle, until confusion finally turns to illumination.

• **You know things that no one else does.** Never let insecurity restrain you. Good ideas come to fools and saints alike, and expertise is no guarantee of quality (sometimes quite the opposite). To judge good ideas from bad, there's only one way: You must give the idea a chance. More than one idea that *seemed* stupid on the face of it turned out terrific or led to some terrific place. Real-world example: Post-it notes. ("A glue that *doesn't stick*? Come on!") Recorded example: Tears For Fears' "Everybody Wants To Rule The World," which started as a studio joke and turned out to be their big break in America.

Buy yourself a notebook. Go nugget-hunting and keep tally of the score. Track every idea you try, but from three different perspectives: Your first hopes, your immediate gut reaction, and your feelings on the following day. Don't rush to judgment or you'll miss things, just like the hotshot Hollywood exec who dismissed Fred Astaire's first screen test with the comment, "Can't sing, balding, dances a little."

• **Professional engineers are handicapped by their profession.** It's perverse, but originality requires experimentation, and experimentation requires time, and time (at hundreds of dollars an hour) is often exactly what big-ticket recording projects do not provide.

If you're not working under the studio clock, take advantage of your freedom and experiment like crazy. Read everything you can. Ask *lots* of questions. Don't settle into one approach: Juggle them. Should you find yourself drifting, set a goal—a drum track, a song, an album, trying out all umpteen million available combinations of your new reverb's settings—and stick with it until done. Whatever you do, *don't take shortcuts*. Shortcuts are for people who don't have time to enjoy the scenery.

If you *are* "taking what they're giving 'cause you're working for a living," remember that even pro quarterbacks have to do daily stretches. To engineers, that means improving ears, hands, and breadth of knowledge. For ear training I recommend "blindfold" tests, where one person works the

> "Knowing how to arrange for orchestras doesn't mean you can write for synths. Razor-blade chops don't guarantee digital editing accuracy."

The Electric Near-Field Acid Test

WE AUDITION 14 PAIRS OF MONITORS IN A
COMPLETELY SUBJECTIVE, BLINDFOLD AURAL EXAM.
WOOF!

SELECTING MONITORS for the near field is one of the most important decisions to be made when you're putting together a recording studio. In smaller rooms, near-field monitors typically provide the single tonal reference, due to space and monetary constraints. Even in larger facilities, engineers and producers tend to prefer them for mixing because the direct nature of their sound is less colored by room acoustics. And regardless of a studio's size, near-field monitors offer a closer representation of the average speakers found in real-world listening environments.

The near-field monitor market has expanded greatly, thanks to the growing number of personal studios. Many manufacturers now produce near-field monitors that they claim are accurate in both

frequency response and depth of field. Yet each monitor system sounds different, and the methods used by manufacturers to come up with specs seem arcane at best. So how can you tell which manufacturers' claims are accurate? Are certain sets of monitors better suited to certain types of music? Are flat monitor systems the most desirable? What's the best way to evaluate the difference between one monitor system and the next?

To address these questions, we decided to take a bulk eraser to the specs and rely on the old ears and grey matter in the first of a series of *EQ* Listening Tests. Admittedly, this makes our evaluation completely subjective! (See "Reader Beware" on page 34.)

We brought arm-load after arm-load of near-field monitors into a room, and in-

BY ROLF HARTLEY

Stereo Imaging

Width Average
Shows perception of left/right stereo imaging width as compared to the control monitors.

Solo Piano
Dance
Rock

Depth Averages
Shows perception of sonic depth-of-field as compared to the control monitors.

Solo Piano
Dance
Rock

Dispersion Averages
Shows perception of directionality as compared to the control monitors.

Solo Piano
Dance
Rock
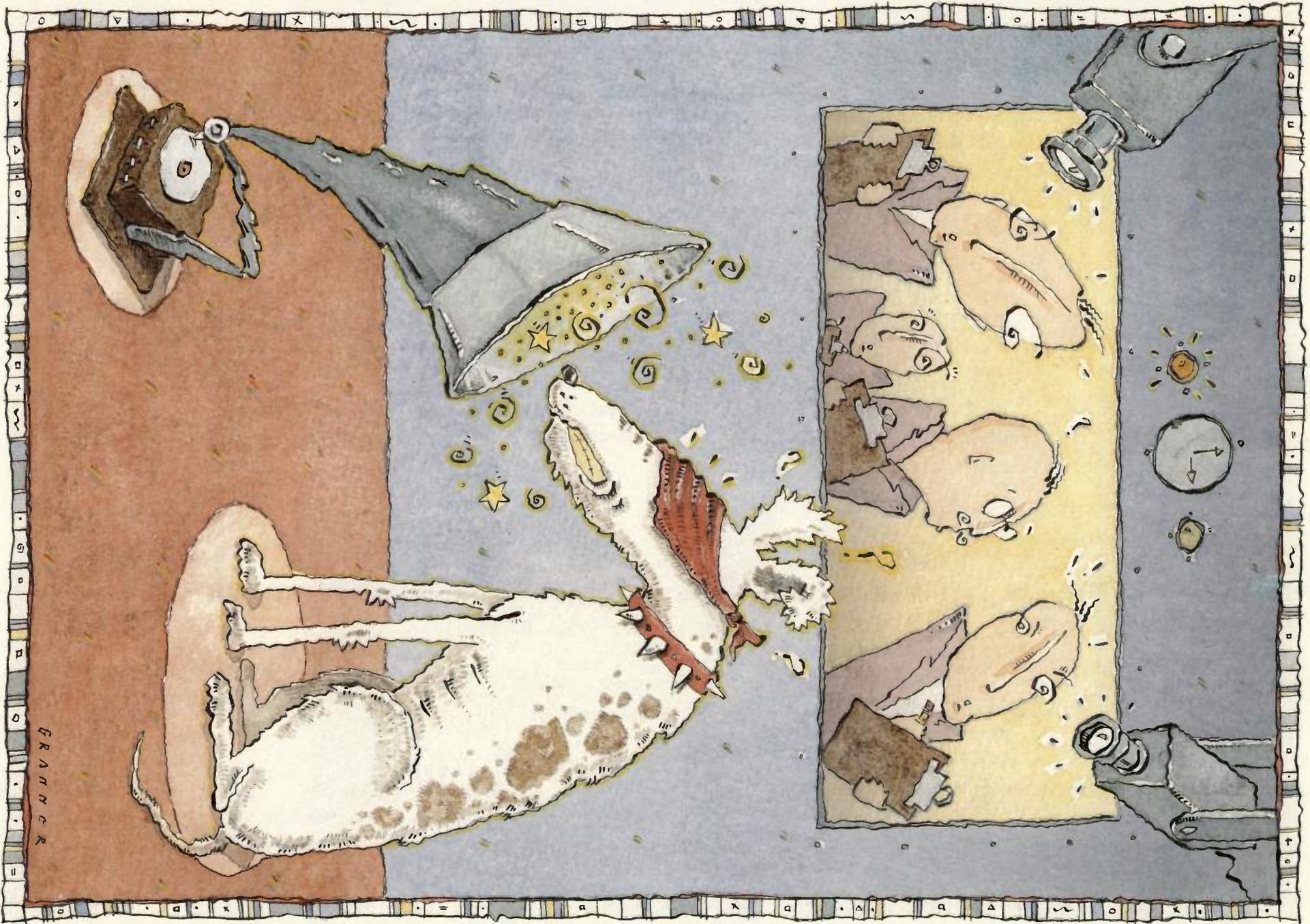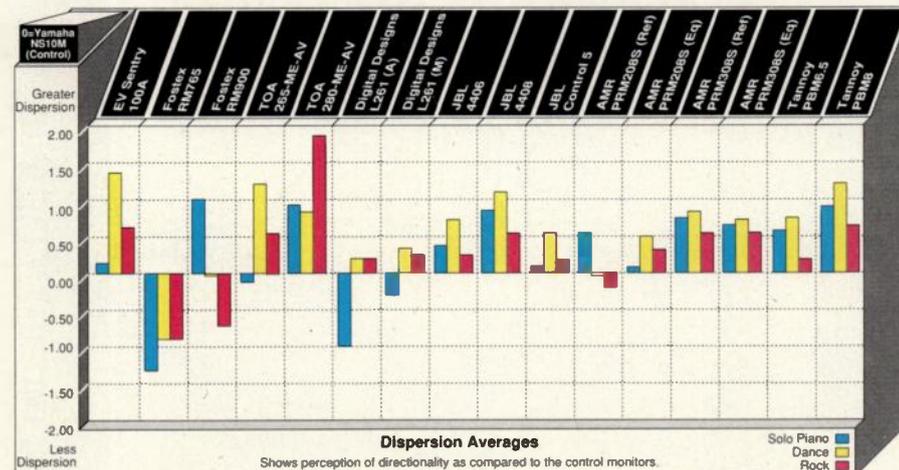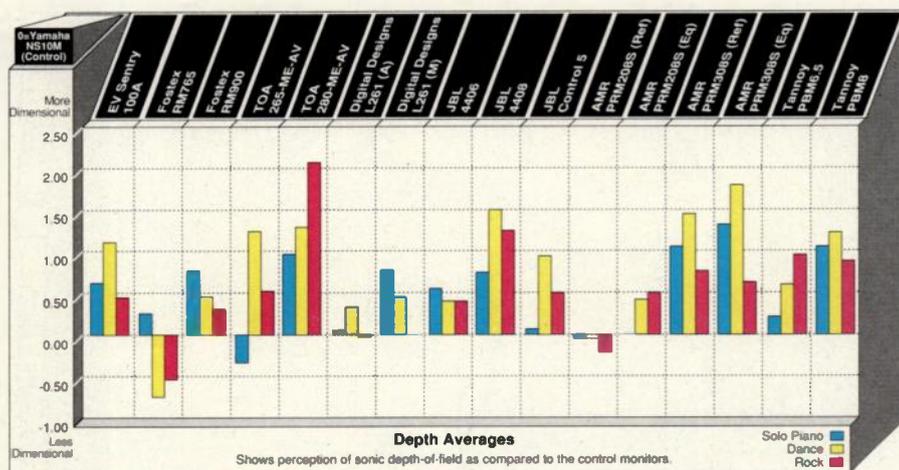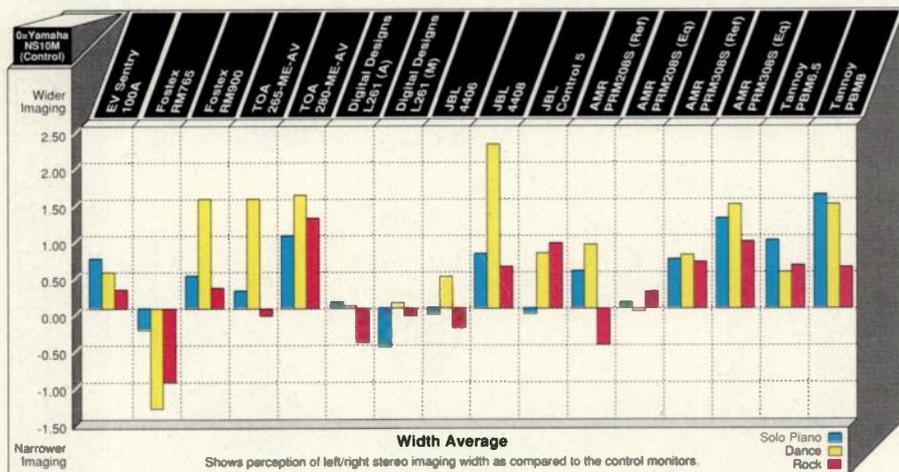
vited a group of ten professional engineers and producers (eight men, two women) to compare them without knowing which models they were hearing. Besides offering some interesting verdicts, the tabulated results provide a guide to help in your own listening comparisons.

The Test Monitors. Our premise for testing was based on several criteria. First, we excluded any monitor that retailed for over $1,000 per pair, since we felt this was the financial limit of the "average" end user. As it turned out, the average cost of our monitor subjects was in the $400-500-per-pair range. Second, since we were testing for the near-field environment, we decided to confine bass driver size to no smaller than six inches and no larger than eight inches. Thus our evaluation covered speakers, marketed as "studio monitors," that fit these basic criteria.

We chose 13 sets of monitors to compare to a set of control monitors. Yamaha NS10Ms ("Studio" version) served as the "control pair," since they are recognized as an industry standard in near-field monitoring. Although our control choice may be controversial, the NS10Ms are used internationally and therefore invite a natural comparison with the new breed of near-field monitors.

Three sets of test monitors were tested twice because they featured selectable EQ curves. These monitors were designed to be switched between a "popular" response curve and an EQ curve optimized for that monitor's components and enclosure.

Power for all monitors was provided by a Crown Micro-Tech 600LX (rated power of 135 watts per channel into 8 ohms). We used the dbx DX5 compact disc player to play the music selections because its loop mode allowed the same musical passages to be repeated indefinitely while we switched between monitors. Monster Cable supplied custom speaker cable harnesses made of their MCP Prolink cable.

A Switchcraft switcher was used to compare the speakers. No automatic level adjustments were made to compensate for speaker efficiency differences.

(This was a hard decision, but one made since our experience with level-compensated switchers is that they induce all kinds of nasty phase distortion. Admittedly, the lack of automatic level compensation creates the weakest objective link in our test, since "louder" often is perceived as

"better." All listeners were told to be aware of this, and were encouraged to audition at a wide range of listening levels.)

The Ears Have It. It became apparent that if the tests were to be as fair and impartial as possible, we would have to create a "blindfold" listening environment to nullify any predispositions and biases towards a particular product or manufacturer. We stretched ultra-sheer black nylon across the front of the monitor platform and removed the speaker grills to reduce interference. Then, by brightly back-lighting the monitors and conducting the listening sessions at night in a darkened room, we made the monitors virtually impossible to identify by sight. (At the end of each evaluation, all the listeners felt confident that they knew which monitors were which, only to be shocked they had chosen a completely different set as their favorite!)

Next, we put a line of masking tape across the floor at four feet from the front of the monitor platform. Two reasons: first, because the near-field theory considers two to five feet as the distance a listener should be from the speakers; second, because the nylon facade was so sheer, if you got close enough to the monitor platform you could identify the products. Only one person at a time took the test, eliminating the pressure of peer influence. The test had no time limit.

Our test considered five main categories for speaker evaluation: "stereo image" (including width, depth, and dispersion of the stereo field), "spectral balance" (including flatness, bass extension, and treble extension), "transient handling," "clarity," and "gestalt." Comments were welcomed.

These criteria were applied to three categories of musical program: solo piano, dance, and rock. We used a 10-point system (-5 to +5) to score better or worse comparisons between each set of test monitors and the set of control monitors. A score of zero represents "no better" or "no worse" than the control monitors.

Monitoring The Results. The charts found on these pages depict the results of our listening test. It is extremely important to understand that each chart represents an average of the scores given by our listeners for that category. At the left side of each chart is the point spread that shows just how much better or worse our group felt about a particular set of monitors. While looking over the charts, remember that the

Spectral Balance

Flat Response Averages
Shows perception of freedom from monitor-induced frequency coloration compared to the control monitors.

Bass Extension Averages
Shows perception of low-frequency handling compared to the control monitors.

Treble Extension Averages
Shows perception of high-frequency handling compared to the control monitors.

Additional Tests

Transient Handling Averages
Shows perception of ability to reproduce highly dynamic attacks as compared to the control monitors.

Clarity Averages
Shows perception of uniform clarity in all frequency ranges as compared to the control monitors.

Gestalt Averages
Shows subjective musical preferences compared to the control monitors.

control monitors are represented in each graph as the zero line.

The final statistics from each evaluation and listening category were averaged into the "All Things Considered" chart. Equal weight was given to each category in the process—perhaps not a realistic approach, but the most manageable.

Besides showing how the products compared, we hope this evaluation sheds some light on the highly subjective nature of monitor evaluation. Everyone has stylistic preferences in music, mixing, and production (and acoustic environment); these ultimately serve as the primary criteria when you're choosing a pair of near-field monitors. Just as the members of our evaluation group drew their own individual conclusions, so must you. •

Monitor Specs

The following frequency response and power handling specs are those claimed by the manufacturers; they have not been tested by *EQ*.

Yamaha NS10M "Studio"
Type: 2-way
Frequency Response: 60Hz-20kHz (± dB not indicated)
Power Handling: 60 watts program; 120 watts peak
Woofer: 7"
Tweeter: 1-3/8" dome
Enclosure: Sealed, mirror-imaged
Controls: None
Price: $475.00/pair

The NS10Ms were chosen as the anonymous control pair due to their industry-wide acceptance. In general, they were not liked by our listeners. They are shy on the bass end, which may cause inexperienced engineers to produce mixes that boom on other systems. To their credit, they *are* the industry standard: Engineers and producers can move from studio to studio and know what they're hearing. Originally hi-fi speakers, the "Studio" system offers better transient handling, yet the monitors are still bright.

Electro-Voice Sentry 100A
Type: 2-way
Frequency Response: 45Hz-18kHz (±3dB)
Power Handling: Continuous (above

40Hz) 30 watts; peak (10ms above 40Hz) 300 watts
Woofer: 8"
Tweeter: 1" dome
Enclosure: Front-vented, vertically aligned
Controls: Shelf-type high-frequency variable adjustment (2 dB boost to 4 dB cut)
Price: $720.00/pair

The EVs fared moderately well, 'though they were perceived to have a lot of boom and "sizzle." While doing very well in the dance music category, the Sentry 100s were felt to be more obscure-sounding in their handling of rock and solo piano music.

Fostex RM765

Type: 2-way
Frequency Response: 70Hz-20kHz (±3dB)
Power Handling: 120 watts continuous
Woofer: 6.5"
Tweeter: 1/2" co-axial
Enclosure: Front-vented reflex
Controls: Field switch to optimize environment response (2π for soffit or wall mounting; 4π for free-standing or console mounting. We tested them at 4π). High-frequency control (2dB boost-12dB cut).
Price: $500.00/pair

The RM765s came out with a negative score in almost every category and test parameter. In defense, it must be said that they are very directional, offering an extremely tight window of stereo imaging and spectral balance. We could recommend these speakers for use in environments that are less than acoustically ideal; the RM765s are so directional, they isolate the listeners from the room, allowing prudent EQ decisions in an otherwise hostile situation.

Fostex RM900

Type: 2-way
Frequency Response: 55Hz-25kHz (± dB not indicated)
Power Handling: 110 watts continuous
Woofer: 8"
Tweeter: 1" co-axial
Enclosure: Front-vented, bass reflex (phase-inverting)
Controls: Field switch to optimize environment response (2π for soffit or wall mounting; 4π for free standing or console mounting. We tested them at 4π). High-frequency control (1dB boost-13dB cut).

The Final Tally

All Things Considered
Shows overall test averages (with equal weight given to all categories) compared to the control monitors.
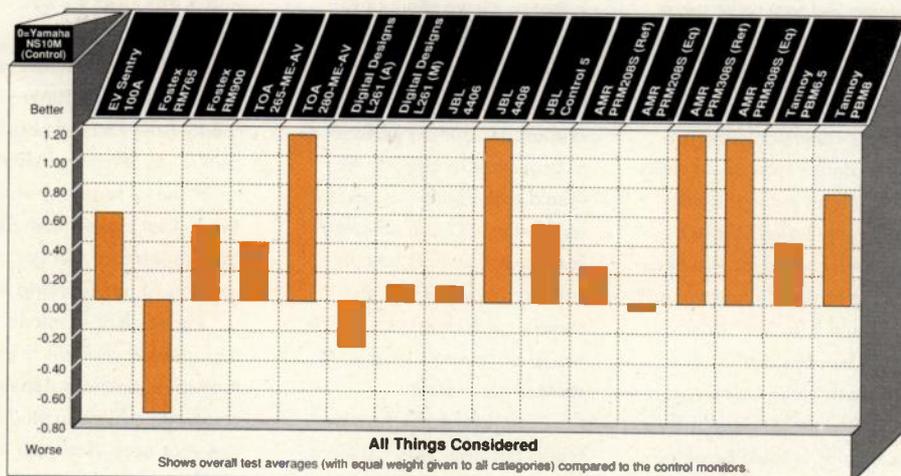
Price: $800.00/pair

The RM900s turned in a better showing than the RM765s, particularly in the solo piano category, with good scores in the gestalt and clarity tests. Overall, the 900s weren't that strong, especially in the rock category, where they scored low in the clarity and flat areas. The RM900s seem better suited for classical recording.

Toa 265ME-AV

Type: 2-way
Frequency Response: 60Hz-20kHz (± dB not indicated)
Power Handling: 75 watts continuous; 200 watts peak
Woofer: 6.5" polypropylene
Tweeter: 1.2" dome
Enclosure: Front-vented, bass reflex, mirror-imaged
Controls: None
Price: $389.00/pair

The "AV" distinction in the model number of this monitor refers to its shielded status, allowing use in close proximity to video monitors, computers, and other devices that are susceptible to magnetic disturbance. These monitors did their best in the dance music categories, garnering third place in the bass and treble extension tests. The 265s did okay in the rock category, but very poorly in the solo piano category, dragging down their overall score. If you're on a budget and writing dance-oriented material, the

265ME-AVs just might be your cup of tea.

Toa 280ME-AV

Type: 3-way
Frequency Response: 60Hz-20kHz (± dB not indicated)
Power Handling: 90 watts continuous; 250 watts peak
Woofer: 8" polypropylene cone
Tweeter: 1.2" dome
Super-Tweeter: 0.8" dome
Enclosure: Sealed, mirror-imaged
Controls: High-frequency level control w/ no markings.
Price: $499.00/pair

The 280MEs came through with flying colors, taking second place overall. They were the best liked in the rock category, proving the highest scores overall in the stereo image area. If you're looking for a good speaker, definitely give these a listen. Like the 265ME-AVs, these also are shielded speakers.

Digital Designs LS261

Type: 2-way
Frequency Response: 54Hz-21kHz (± 3dB)
Power Handling: 200 watts continuous
Woofers: 2 x 6.5" polypropylene
Tweeter: 1" dome
Enclosure: Sealed, mirror-imaged
Controls: EQ switch for "A" (accurate) and "M" (popular mid-range)
Price: $519.00/pair

Reader Beware

When the concept for our Listening Test first came up, it seemed as if there were a million things wrong with it: Whose ears could be "golden" enough? How would the ratings be done? How could every possible fault in the test's design be eliminated? Would there be a control, or would everything have absolute ratings? How could it be fair? How could it be objective?

It wasn't long before we realized the terrifying truth: There was no way it could be objective. There was no way every fault could be removed. There was no way it could be perfect.

What this said, instead, was that we were about to undertake a completely subjective test—a test that would be, in some respects, as invalid as it would be valid. We would do our best to eliminate variables. We would choose a range of listeners, all of whom had years of studio listening experience. We sought a mid-sized (18'x25') room with carpeting and drapes, but no professional acoustical treatment. We would make hard choices, such as whether to have automatic level compensation (and its sonic drawbacks) with the switcher (we chose not to). We would upset some people. We went ahead with it.

During the testing, some things became clear. Listeners generally agreed with each other—and they did so without the bias of knowing how the others felt. It was easy to hear definable differences among many speakers. At low- to moderate-level direct listening, we felt the room was not a major influence; if anything, its meager acoustic treatment was representative for many users. And it was clear we had a test that meant something.

But it doesn't mean everything. Yamaha NS10Ms—the control pair—scored poorly. Yet they're still *the* near-field monitor of choice for many engineers. That makes them an excellent *reference* monitor, regardless of our Listening Test.

Above all else, remember that speaker choice is the most subjective purchase decision you'll make for your studio. Use your ears. And your heart.

If you love the sound of a pair of monitors, and your research concludes that they will do the job they need to do, buy them. Don't let anyone else tell you otherwise.

—*Brent Hurtig*

We tested both EQ settings. In the "A" position, these speakers were not liked, turning in the second-lowest scores after the Fostex RM765s. They were particularly weak in the clarity category for each musical selection. The "M" position proved somewhat better, hovering pretty close to the zero line in most categories. You could think of the LS261s with the "M" setting as NS10Ms with a little better bass. Overall they were disappointing, though if you really like NS10Ms, you may love these speakers. [*Ed. Note: Since our Listening Test, the LS261s have been slightly modified. We're told units now shipping have a slightly increased output over our test pair.*]

JBL 4406

Type: 2-way
Frequency Response: 55Hz-20kHz (± 2dB)
Power Handling: 75 watts continuous
Woofer: 6.5" polypropylene cone
Tweeter: 1" diaphragm
Enclosure: Front-vented, non-mirror-imaged
Controls: High-frequency control with no markings.
Price: $470.00/pair

The 4406s scored pretty close to the zero line in most categories, showing best in the dance category. For their size, the 4406s really kick bass. These would be an ideal set of monitors for people who need extra bass extension but don't have the space to devote to a larger speaker.

JBL 4408

Type: 2-way
Frequency Response: 50Hz-20kHz (± 2dB)
Power Handling: 100 watts continuous
Woofer: 8"
Tweeter: 1" diaphragm
Enclosure: Front-vented, non-mirror-imaged
Controls: High-frequency control
Price: $570.00/pair

The 4408s pulled a strong fourth place overall, doing their best in the dance and rock categories. No negative scores were posted concerning the 4408s; however, their lowest scores were found in the "flat" test. This indicates these monitors could stand improvement in their tonal balance characteristics (the middle frequency ranges particularly seemed a bit subdued). The 4408 is a great, smashing, rock 'n' roll speaker.

JBL Control 5

Type: 2-way
Frequency Response: 75Hz-20kHz (± 3dB)
Power Handling: 175 watts continuous
Woofer: 6.5"
Tweeter: 1" dome
Enclosure: Front-vented, non-mirror-imaged
Controls: None
Price: $375.00/pair

This system has A/V-shielded magnets so it can be used in close proximity to video monitors and other devices that are susceptible to magnetic disturbances. The Control 5s turned in a pretty good showing at seventh place. For their size, the Control 5s were particularly smooth and were thought to have a good frequency response, although a bit light in the midrange. Particularly strong in the dance music category, the Control 5s were considered punchy and clear-sounding. They could find a strong foothold in home entertainment systems.

Audio Media Research (AMR) PRM208s

Type: 2-way
Frequency Response: 68Hz-18kHz (± 3dB)
Power Handling: 60 watts program; 30 watts continuous
Woofer: 8"
Tweeter: 1" dome
Enclosure: Sealed enclosure, vertically aligned
Controls: Switch between REFERENCE (flat) and EQUALIZED (optimized) EQ settings
Price: $478.00/pair

This monitor was tested twice, once for each EQ setting. The REFERENCE mode provided a usable sound though noticeably light in bass response. The high end was smooth and clear. Overall, you can consider the PRM208 in REFERENCE mode as a darker NS10M with a more coherent midrange.

In the EQUALIZED setting, the PRM208s were disappointing, proving the lowest of all scores in the clarity and gestalt areas of the solo piano category. For rock and dance, the 208s were no party either,

coming in at third-to-last overall.

AKR PRM308s

Type: 3-way
Frequency Response: 45Hz-18kHz (± 3dB)
Power Handling: 80 watts program; 40 watts continuous
Woofer: 8"
Mid: 5"
Tweeter: 1" dome
Enclosure: Front-vented, mirror-imaged
Controls: Switch between REFERENCE (flat) and EQUALIZED (optimized) EQ settings
Price: $599.00/pair

We also tested this monitor for each of its EQ settings. The EQUALIZED mode for the PRM308s was very strong in the rock and dance music categories, proving the highest scores in the gestalt, clarity, and depth areas of the dance music categories. On top of these honors, the 308s were felt to have some of the best bass extension of any monitors tested. In the EQUALIZED mode, they are smooth and provide a very deep stereo image that must be heard to be appreciated.

In REFERENCE mode, the PRM308s took first place as the best-liked speaker of any tested. In this mode they were thought to possess the clearest and most musical sound for all three types of music. The main difference between the two EQ settings lies in the frequency balance characteristic, where the REFERENCE mode is much more even sounding than the EQUALIZED mode. Many of the near-field monitors tested had a light midrange feel to them, so it's nice to be able to flip a switch to get an idea of how your music sounds in an alternate environment. The PRM308 system is truly wonderful and must be heard to be believed!

Tannoy PBM6.5

Type: 2-way
Frequency Response: 57Hz-20kHz (± 3dB)
Power Handling: 125 watts
Woofer: 6.5" polypropylene
Tweeter: 3/4" dome
Enclosure: Rear-vented, vertically aligned
Controls: None
Price: $349.00/pair

Despite their small size, PBM6.5s are really strong in the bass department, having one of the best bottom-ends tested.

A Monitor Glossary

Bass extension. The relative degree to which a monitor is able to reproduce low frequencies. Pronounced bass extension is not necessarily an indication that a speaker's response is not flat.

Clarity. The detail with which a speaker is able to reproduce individual sonic elements within a mix.

Co-axial. A type of speaker manufacturing design in which the high-frequency driver is mounted in the center of the woofer.

Depth. The perceived sonic depth of field.

Dispersion. The directionality with which a speaker radiates sound. In practical terms, the width of coverage from the mid- and high-frequency drivers dictates the width of the speaker's dispersion. Narrow dispersion confines the listener to a narrower "sweet spot" for optimal listening.

Flat response. In a monitor, the ability to reproduce audio signals without inducing any frequency changes to the signals entering the monitor. Unwanted changes in response are known as colorations. Even in a test chamber, few monitors are able to deliver frequency response much flatter than 40Hz to 20kHz (±6dB).

Gestalt. A purely subjective gut reaction or emotional response (to the overall sound of a monitor, in this case).

Mirror imaging. A design by which monitors are specifically designated as the left or right elements of a stereo pair.

Near-field monitor (NFM). A loudspeaker designed for up-close, low- to moderate-level listening. Most NFMs are designed for placement within one meter of the listener.

Sealed. An airtight cabinet design with no vent or port (see "Vented").

Spectral balance. The balance of tones across the entire musical frequency range.

Stereo image. A term generalizing the placement of sounds within the stereo listening field. Width, depth, and dispersion are characteristics that contribute to the overall stereo image.

Transient handling. The ability to reproduce highly dynamic attacks—such as those from a kick drum or a rim shot—with accuracy.

Treble extension. The relative degree to which a monitor is able to reproduce high frequencies. Pronounced treble extension is not necessarily an indication that a speaker's response is not flat.

Vented. A speaker cabinet design that features an open, tuned port to allow the flow of air caused by low-frequency driver movement to escape.

Width. The perceived spread of the sound field between a left and right speaker.

Unfortunately, the rest of the speaker can't keep up with the bass, for it possesses a somewhat dark sound. The PBM6.5s would be a good system for engineers plagued by bass-heavy mixes, because it feels as if there's more bass there than there is. The PBM6.5s are also a nice choice if your work requires you to haul around a set of monitors—they're mighty compact.

Tannoy PBM8

Type: 2-way
Frequency Response: 47Hz-20kHz (± dB not listed)
Power Handling: 125 watts
Woofer: 8" polypropylene
Tweeter: 1" dome
Enclosure: Rear-vented, vertically aligned
Controls: None

Price: $479.00/pair

The PBM8s turned in a decent showing overall, holding their own in the rock and solo piano categories. The dance music portion of the test was not as kind to the PBM8s, because they were considered too dark-sounding for this category. Primarily strong in the stereo image group of the solo piano category, they possess a round, warm tone for that type of recording. If your work involves a lot of detail- and ambient-listening, the PBM8s are a fine choice. •

EQ would like to thank all the manufacturers for their assistance. Special thanks to Monster Cable for the cabling.
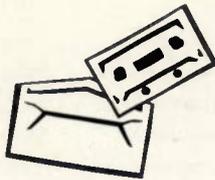
SELF·MARKETING

To Market, To Market

BY JOHN SUGNET

MANY SMALL-BUSINESS PEOPLE tend to look on marketing and advertising as necessary evils, or worse, as if they were dragons that must be appeased by a bloodletting of dollars every new moon. While it may be true that marketing is not an exact science, it is a craft that blends rational analysis and creativity.

As a recording professional, you already are involved in a similar form of alchemy. And you can re-channel some of the same creative energy you use in making great recordings to get better clients, higher profits, and more interesting projects. Let's start out by looking at some basic marketing concepts that are prerequisites to planning any successful campaign. You can apply these principals to market anything from freelance services to a world-class studio.

Make Yourself The Client. Marketing, like any other project, requires time and attention to detail. Set aside a couple of hours each week to meet with yourself and plan your marketing. Begin by scrutinizing your business the same way that you would examine material someone brought you to record. Before you can market your business, you need to have a clear picture of what it is you have to offer.

Get feedback from people who have used your services about what they liked and didn't like. Evaluate your strengths and weakness, in terms of services, facilities, abilities, and experience. Remember to include outside talent and equipment that are available to you on a contract basis—they can be important resources.

Target Your Market. Once you've taken stock of what you have to offer, you'll need to evaluate your services within the context of the marketplace. Develop a realistic list of market segments—types of clients who might use your services.

If you are uncertain about whether your facilities are adequate for handling a particular kind of project, get on the phone and find out from facilities that already are handling those projects. You'll find most people willing to give you some helpful information.

Prioritize your list of market segments, beginning with the segment whose needs most closely match the services you offer. Since it is *people* rather than "market segments" that you want to communicate with, build a personal profile to represent each market segment you

want to reach. It's difficult and unnerving to try to communicate with a vast unknown public. It's much easier to talk to a person.

Imagine a heavy metal rocker who wants to make a basic demo in order to get club dates, and a new age musician looking for a place to do a low-cost, independent album project. Both might be potential clients, but they have very different profiles. They speak different languages, read different publications, and have different tastes and concerns.

Developing personal profiles for the various market segments that you want to reach will help you determine *what* you want to say, *how* to say it, and the *most effective medium* in which to get your message out.

Sell The Benefits. Ultimately you are selling your clients benefits, not services. You and your competitors may have comparable equipment, and talent, but there are benefits that are unique to your company. Such characteristics as faster service, more pleasant atmosphere, better location, better understanding of the client's needs, more reasonable rates for the same services, more flexible hours, and personalized service combine to make up the total benefit package.

When communicating with potential clients, it's important to carefully consider their particular needs, then stress the benefits you offer that meet those specific needs. Small things can make a big difference.

Position Yourself. In a major market where potential clients can choose from a broad range of services, it is especially important that you develop a clear identity. You'll need to let the client know why she or he wants *your* services.

The "we try harder" campaign by Avis is a classic example of good market positioning. By identifying itself as the "number two" provider of car rental services, and stressing the idea that this made Avis employees work harder, Avis turned its position as second-best into a benefit for its customers.

Occupying a unique position in the client's mind means stressing what is different about your services in a fresh and interesting way. If your message is too general, it won't be memorable. Speak to the needs of the market segments you're best able to serve. Don't try to be all things to all people.

Develop Your Image. Every great song has a strong hook, and every great business has a strong image. Everything about the way you present your business, from the way you answer the phone to the look of your invoices, says something to the client about your company. Your task in developing an image is to condense the benefits you offer to the client into a concise idea, and to express that idea in a unique and exciting way.

A great business image is more than just slick graphics or clever words. It is the ultimate expression of the personality of a business.

> "Occupying a unique position in the client's mind means stressing what is different about your services in a fresh and interesting way."

IN THE POCKET

Your Ticket To Business Savvy

BY STEVE HARLAN
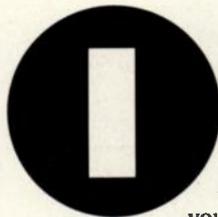
IF YOU'RE LIKE MOST studio owners, engineers, producers, composers, and other creative types, the chances of your enjoying the tax and accounting end of your business are about the same as the chances of your calling the IRS and asking them to come out and audit your last five income tax returns.

But business, as they say, is business. Doing it right can make the difference between success and disaster. That's the reasoning behind "In The Pocket."

In the coming months we'll discuss a multitude of tax, business, and financial planning ideas. We'll attack the subject in a way that (hopefully) takes the pain out of the absorption process. Being a creative studio type myself, and having a few of them among my clients, I have a pretty good idea about when a conversation might start making eyes glaze over. But believe it or not, this business stuff actually can be interesting, or I wouldn't be doing it myself!

To assist you in the acquisition of business savvy—and to keep more of your hard-earned dollars in your pocket—here are a few topics I'll cover in future issues of *EQ*:

Different Ways To Organize Your Business. The legal form in which your business operates can make a big difference. Should you incorporate? How about a partnership? What's involved and how much will it cost to set up? This primer will familiarize you with concepts and terminology so you won't be bamboozled when your attorney asks you, "Will this be a section 351 incorporation with a 1244 election?"

Saving Tax Dollars With A Subchapter-S Corporation. If you know what self-employment tax is and know you don't like paying it, you can lighten the load with a salary/rent/distribution compensation setup with a subchapter-S corporation. I'll tell all.

How To Write An Effective Loan Proposal. Maybe you've dealt with lending institutions in the area of home and auto loans, or perhaps financed some minor equipment purchases. But if you've ever approached a lender for the purpose of financing a major business equipment purchase or studio improvement, you know you're in a whole different arena. Understanding what makes an effective loan proposal can separate you from the amateurs. We'll discuss what bankers look for, and even gain some insight into how they think.

Independent Contractors. Do you pay for subcontracted talent in your business? Are you aware of the legal distinction between employee and independent contractor? Did you know the IRS is aggressively going after businesses that improperly classify employees as independent contractors and fail to withhold taxes? We'll talk about the rules and offer some tips on structuring independent contractor agreements so they stand up under IRS scrutiny.

Qualified Retirement Plans—Beyond IRAs. Qualified retirement plans remain one of the few good tax shelters under tax reform. Did you know you can put up to $30,000 each year into a qualified retirement plan, *tax free*? If you're serious about starting that nest egg, this article will be required reading.

Production Projects: Estimating A Fee That Will Earn You A Profit. Have you ever taken on a project for a client who demanded a "not-to-exceed" fee, and the fee you quoted ended up earning you less than a pauper's wages? I think we've all been down *that* street. Estimating project costs is more an art than a science. We'll discuss establishing a "project cost database" and a checklist to be used in the estimating process. We'll talk about situations that are candidates for a solid estimate, and situations where you'd be crazy to quote anything other than an hourly rate plus hard costs.

Work Scheduling: How To Meet Deadlines And Retain Your Sanity. In the recording, production, and composing business, we often are called upon to meet impossible deadlines. Developing an information system that takes away some of the guesswork is a must.

Solid reputations are built when quality soundtracks are delivered on time. Committing to deadlines under pressure from clients—then finding you have no chance of meeting them—ranks you with the amateurs. A systematic approach to work scheduling will prevent you from agreeing to unreasonable deadlines and help you meet the other kind.

To sum it up, when it comes to dealing with business and taxes, a lot of creative types feel like fish riding bicycles. Awkward, if not unnatural. To combat this feeling of business inadequacy, they'll buy business books, take business courses, and so on. Although this is a good idea, unfortunately these books and classes often have little practical application to the business situations in our creative industries. "In The Pocket" will talk about your business, in your language. So get ready to increase your business savvy and put some distance between you and the competition. •

> "In the recording business, we often are called upon to meet impossible deadlines."

SELF-MARKETING

Many small businesses try to take on a "classy" or trendy image that has no organic basis. They end up looking stuffy and insincere.

When choosing a name, logo, or slogan for your business, make sure it sets a tone that reflects *who you are* as well as what you do. Don't let any "experts" or artists talk you into something that doesn't feel comfortable.

Invest plenty of time in tinkering with your image. Bounce it off friends and give their feedback time to sink in. Shoot for simplicity, and don't be trendy: You want your image to be around for years to come. Make sure any artwork you develop works well in all media, color or black-and-white. Once you have established a look, be consistent. Ads and all collateral material should present one image.

In the next issue we'll look at executing your marketing strategy—the specifics of putting together a campaign. •

WHITE NOISE

gear while the other analyzes what is being tweaked, and by how much. For limbering up hands, nothing beats speed trials—ten to 15 five-minute mixes of the same material, as varied as possible, with only enough time between runs to rewind the tape and clear the board. As for increasing knowledge in a *useful* way, that's easy. We all have gaps in what we know. Whichever gap we'd be most embarrassed to admit is the one we most need to fill.

• **New Situations Make Everybody A Beginner.** Twenty years of analog experience don't prepare you to become a sampling fool. Knowing how to arrange for orchestras doesn't mean you can write for synths. Razor-blade chops don't guarantee digital editing accuracy. Knowing MIDI inside and out won't help you record a killer backbeat. These are matters determined by personal talent, ambition, smarts, and perspiration (as in what Thomas Edison said genius was 90 percent of).

It all comes back to Charlie Chaplin, and down to this: To do something in this world, you have to be the best damn beginner you can be. •

EVERYBODY MUST GET ZONED

L.A.'S STUDIO CONTROVERSY MAY AFFECT ALL OF US. HERE'S THE UNTOLD STORY. BY CARY TENNIS.

It was a swell Los Angeles ranch house with no toys on the lawn and no cars up on blocks. Parked on the street were a Porsche, a Volvo, and a Detroit sedan of the type that a guy named Frank from Mutual of Omaha might drive.

Through half-closed venetian blinds in the front window I could see a young woman sitting at a desk. She was on the phone. Her Macintosh computer was on. She saw me and waved me around to the side of the house. I walked back there. At the rear of the house was a fence. Behind the fence was a barking dog.

I'm afraid of barking dogs, so I returned to the window. "Dog," I mouthed to her.

"It's okay," she mouthed back, and again waved me around the house.

They must take me for a musician, I thought: a customer.

I went back to the fence and stood there. I wondered if it was really okay to just drop in. The *Billboard* studio directory where I'd found this address said it was a "major facility for the production of prerecorded music." It looked to me like a nice three-bedroom, two-bath with a fence, a yard, and a barking dog.

MY ASSIGNMENT WAS TO EXPLORE the culture of home studios in Los Angeles. There was a fight going on. On one side was the Hollywood Association of Recording Professionals (HARP), a group of major studio owners who pay their payroll taxes and pay for commercial space and pay for commercial insurance and pay for commercial phone lines and pay for everything else a business owner pays for. On the other side was the growing number of musicians and engineers who had studios in their homes, who handled mostly private projects but sometimes appeared to operate like full-fledged studios. The home studios didn't play by all the rules the "big boys" played by, and didn't pay all the costs.

The simmering conflict exploded over a guy named Chas Sandford, who, suffering either a lapse of judgment or a fit of monumental chutzpah, issued a glossy brochure advertising his home studio. A member of HARP got a hold of it, brought it to a meeting, and the next thing you knew, city officials had shut down Sandford's studio. HARP denied that its members tattled on Sandford, but tempers flared on both sides. "Unfair competition," claimed the major studio owners, "he deserves to be shut down." "Mind your own business," replied the home studio owners.

I had tried to piece together the story from my home base in San Francisco. Each side had lawyers who were varying degrees of unfriendly. Nobody really wanted to talk on the phone. It was nearly impossible to get zoning information out of city officials. I faxed HARP's attorney a set of detailed questions. He faxed me back a neat "no comment."

So I flew to L.A., shacked up in Venice, rented a Mitsubishi Mirage, and here I was standing at a six-foot fence in a nice suburban neighborhood with a dog barking at me.

"Get in here. Come on!" came a man's voice. I heard a sliding glass door open. He shooed the dog in and closed the door. Then the gate opened and he stood in front of me. "Hi," he said. "We're out back."

It was as I surmised: He made me for a player. He turned, like I was supposed to follow him. I stood there. He turned back around. He looked down at my briefcase.

"Hi," I said. "I'm a writer." I stuck out my hand. He looked at me again. Then he said, "Well, we're in a session. Why don't you talk to Lynn?"

"I'd like to. She waved me back here."

"Oh. She thought you were one of the guys."

When you're a writer, you're never one of the guys.

> "**If** you're at home writing a novel or painting a painting or editing a film, and making money off it, it's illegal, and if someone lodges a complaint, you can be shut down by Building & Safety."

"WE'RE PERFECTLY LEGAL," his wife and office manager Lynn said when I told her I was writing about the studio wars. "Because we don't rent the studio out." I nodded and smiled but I knew better.

I knew better because I had spent the better part of a gorgeous, nearly smog-free Los Angeles morning at Van Nuys City Hall, Zoning Information. The studio's address was zoned R1. I said to the guy at Zoning, "That's one-family residential, right?"

He said, "Right."

I said, "So you couldn't have a business there?"

"Absolutely not," he said. "Not there." He looked shocked that I would even ask such a thing.

I was running his words through my head while Lynn was explaining to me why they were perfectly legal. "We do all our own work here," she said. "My husband does all his work here. It's just like if I were to do word processing out of my home."

A home word-processing business, according to the present Los Angeles zoning code, also would be illegal.

FOR TWO DAYS I DROVE AROUND L.A., looking at studios listed in *Billboard*'s directory. I also spent a lot of time taking numbers and getting in line at zoning information desks, to verify for myself what was obvious: Lots of these supposedly "major facilities" were in people's houses. It was beginning to seem incredible to me that so many people would be working in such blatant violation of zoning laws.

I finally got in touch with associate planner Patricia Ialongo of the Los Angeles Planning Department's Code Studies department. She is drafting new zoning laws that would allow home occupations as long as they meet certain requirements. She put me straight. What I read in the zoning code book was the truth: Technically speaking, all home occupations are illegal in Los Angeles except for a limited number of doctors, dentists, and ministers. If you're writing a novel or painting a painting or editing film, and making money off it, it's illegal, and if someone lodges a complaint, you can be shut down by Building & Safety.

"So why don't they shut all these people down?" I asked her.

"Building & Safety can't prove anything," she said. "Even people coming and going at all hours—they could be friends."

Well, that was kind of silly. Even *I* could prove that these people are making money off these studios. Anybody with any investigative skills could prove it. The apparent truth is that Building & Safety has neither the motivation nor the personnel to close down all the home occupations in Los Angeles.

Attorney Terrence Everett is a spokesperson for Homeworkers, a group of musicians, studio owners, writers, editors, and others seeking a legal way to continue

working at home. Homeworkers recently formed in direct response to the studio controversy. The problem, as Everett puts it, is that "the city's own goals are in conflict with its laws." Los Angeles has a clean air plan which sets forth air quality objectives for the L.A. Basin and proposes ways to achieve them. Stopping people from driving around is one of the ways the plan endorses to achieve cleaner air. Mayor Tom Bradley himself has endorsed the idea of people working at home. So, in principle, the city is in favor of home occupations, but current law stands in the way.

That is why new law is being drafted.

Ialongo's draft of the new zoning laws will go from Code Studies to the

Don't You Feel So All Alone

To all outward appearances, a home that houses a large recording studio might look just like any other residence. But appearances are deceiving. When high-tech moves into the home, a whole host of laws comes into play—from building codes and public safety to taxes and employee rights—that go way beyond the issue of zoning or right to work at home.

First, installing a professional multitrack studio in a good-sized house frequently involves building code violations: Studio owners who can't get permission to operate their business in a residential area are unlikely to seek building permits before they begin knocking out walls and floating floors.

In addition, the electrical power requirements of a major studio immediately raise concerns about safety. When you start installing 250- to 300-amp wiring systems in a house, the potential for problems increases. How many large home studios, one wonders, have been properly inspected, or carry the necessary amount of fire insurance?

Then there are other issues related to governments. If commercial studios are the losers when business moves into home demo facilities, what about the taxman? Cities and states rely on revenues from a vast array of licenses, special fees, and taxes aimed at businesses. Pro studios shoulder a heavy burden in real estate taxes alone, relative to their home studio counterparts.

What it means is that big commercial studios spend a lot of money that underground home studios do not. Terry Williams of L.A.'s Lion Share Recording Studios (and HARP) is one of those who has taken the lead in addressing the issue. His group has made recommendations to the local government, which is drawing up a new Home Occupation Ordinance. But his goal is not to put competitors out of business. "All I'm asking for is a level playing field. If they [home studio owners] don't have to pay taxes and I do, that's not fair. If they're not zoned commercial and I am, that's not fair. A lot of people [commercial studio owners] don't want to talk about this issue, or they feel that it won't affect them in the long run. Certainly Lion Share has suffered from the publicity this issue has generated, but not to the degree it would if nobody did anything. Because if we don't do something about it, then this issue will resolve itself. In ten years your magazine won't have to write about whether studios like mine are losing business to the home operators, because there won't be any Lion Shares left."

—Phil Hood

Planning Commission, which will hold public hearings. Then a version will go to the City Attorney, who will do some legal fine-tuning, and then it will go before the Los Angeles City Council for a vote. Home occupations in Los Angeles could be legal by this time next year.

WHAT IS LEGAL NOW IS "HOBBY USE." You can get a building permit to build a recording studio in your home if you say it's for hobby use.

I went to see this guy with a home studio out in Sherman Oaks. For a living he arranges and writes synthesizer programs for people like Julio Iglesias and Frank Sinatra. He produced music for the *Dirty Dancing* soundtrack. He's worked with Christopher Cross and Trini Lopez. He's got a digital Mitsubishi 32-track recorder in his home. But he got the permit to build his studio by saying it was for hobby use. That's kind of funny. His *hobby* is flying airplanes.

It was 7 P.M. on the dot and I had an appointment with him. His wife answered the door and said he wasn't home yet. The Chilean-born and -raised producer, composer, engineer, optometrist, former body guard, chauffeur, cook, carpenter, visionary, and pilot was up there right now zooming around.

In the kitchen there was a Girl Scout meeting going on. His wife led me to the studio out back. The studio was very beautiful. I was enjoying the sight and making some notes when he arrived. He showed me how he put the power amps inside the air conditioning intakes so the ducts do double duty cooling the equipment. He showed me how the MegaMix system attached to his computer remembers all the fader positions and replicates their actions via VCAs. He played me some music he did for the Jacksons. Then we got around to the studio wars.

"It's creativity," he said. "You can't zone creativity. I could be sitting on a toilet and writing a song. Should I have to be in a commercial zone?" But on the subject of Chas Sandford, that home studio owner who printed up a brochure and got HARP riled in the first place, he was adamant.

"He should have been shut down. He should not have made that flyer."

By the time I was ready to leave, the Girl Scout party was over. As I walked through the house past the Steinway grand, I could hear the kids splashing in the tub.

I MADE ONE MORE STOP THE NEXT DAY. Michael works in the quality control department of A&M Studios, one of Los Angeles' largest and most respected recording facilities. They employ 45 people; the traffic department uses six Macintosh computers just to keep track of schedules and equipment. Michael works there in the day, but he's got a little studio at home, where he's working on his second album. He gave me a tour of A&M in the daytime, and invited me to see his home studio that night.

I parked the Mirage at 6 P.M. and went through a gate into a courtyard surrounded by Spanish-style garden apartments. Lying in the spotty grass were a child's tricycle, an inflatable whale, a tiny orange plastic chair, a blue boat, and a little yellow gravel-loader made of steel that showed through where the paint had chipped. Hibiscus and palms grew in the courtyard. Through a screen door came the sound of George Bush talking about the Berlin Wall. A baby was crying. There was traffic noise and an airplane flying overhead.

Michael appeared and led me through a small living room loaded with CDs, a Gibson acoustic guitar in a stand, a futon couch. "Here's the studio," he said. It was a little space cramped with a Tascam 8-track tape machine, 24-input board, miscellaneous rack gear, and guitars. "This was the bedroom."

"Where do you sleep?" I asked.

"I sleep out there," he said, meaning the living room. "I don't have a bed—I spend all my money on this stuff. I picked a slummy apartment on purpose, so I could build inside it and the landlord wouldn't care."

Incidentally, though he's part of the home studio wave,

> "**He** got the permit to build his studio by saying it was for hobby use. That's kind of funny. His hobby is flying airplanes."

he's bucking the digital revolution. No synthesizer has ever even been inside his studio. He prefers to beat on metal rather than sample sounds.

"It looks like fun," I said.

"It is," he said. "It's my life."

"**A**RE YOU GOING TO HAVE ME on the cover, biting off the head of Chas Sandford?" asked Terry Williams, who owns Lion Share Recording Studios in Beverly Hills and is a spokesperson for HARP.

"Yep," I told him. "We're working on the graphic right now. Send us a head shot so we can graft it onto the giant lizard body."

Williams was joking, but he is sensitive about being vilified as a big, mean, establishment type. He isn't paranoid, but he is angry and he is worried. He was born and raised in L.A., his parents were both in big bands, and he now runs one of the largest recording studios in the country. He believes it's possible that the big studio era is ending. If that happens, it won't just be the end of a kind of business, it will be the end of a way of life.

What we're seeing is the leading edge of some fundamental, transformative changes in our society, in business, and in the nature of how the musician records music and gets it to an audience. The system used to work like this: To get a record deal, a musician hired a studio, made a demo,

and sent it to a record company. If the musician got a record deal, the label hired a studio and the musician made a record. The musician used the studio's equipment and their people. Stars like the Beatles and Jimi Hendrix built their own studios, but that was an extravagance, on the order of having your own airplane.

Today, even amateur musicians can make quality recordings in their homes. The major decline in studio earnings nationwide over the past year has been in the area of demo recording because musicians are making demos in their homes. They're also making whole albums in their homes. They're doing it in violation of zoning regulations. But zoning laws don't exist to regulate business competition. They exist because people who live in a neighborhood have certain rights. They have the right not to have an auto body shop next door. They have the right not to have somebody build a skyscraper across the street. In

Coming Soon To A Town Near You

Although the pro studio vs. home studio conflict has gone public only in L.A., it soon will appear in a town near you. Reports in a recent issue of *Screen* magazine (a Chicago weekly covering audio and video production) detail home recording's devastating effect upon the demo business in America's jingle capitol. In other cities, some commercial studios are gearing up to protect their rights against commercial facilities that operate from homes. Reportedly, there has even been an attempt on the part of San Francisco-area studio owners to circulate a "blacklist" of home studios that violate local laws.

In Nashville there has been no outbreak of hostilities, but there is little doubt that home recording technology has changed the demo business there. As songwriter Walter Carter put it when discussing the impact on 8- and 16-track studios, "In general, I'd say places that didn't have a big room have disappeared. Back in the early '80s when I started at Silverline Music [a Nashville publishing house], the publishers were getting small Tascams for the back room and doing demos there, without a drummer. Now all the big publishers have their own studios."

To hear a different viewpoint, I called Tommy Goldsmith, a music writer with the *Nashville Tennessean* who also produces records. He put the whole issue in perspective when I asked him whether commercial home studios were taking business from the pros in Nashville. "It's interesting you ask that," he replied in a soft Southern accent, "because I just produced a session like that in somebody's house last Saturday."

—*Phil Hood*

general, they have the right to retain the character of the neighborhood. Zoning laws are supposed to protect those rights.

Because they work toward maintaining things the way they are, zoning laws tend to favor conservative interests. In a world where the status quo is that everyone goes somewhere else to work, it's natural for the zoning laws to say there will be no businesses in residential zones. And

when an economic group is threatened, it will use whatever means it can find to ensure its survival. But as home computers, modems, fax machines, cellular phones, MIDI, database networks, and other technology make it possible to work at home, in your car, and at the beach for that matter, zoning laws must catch up to the times. And major recording studios must realize that what they really have to sell is service and expertise.

Seen at close range, the studio war could look like nothing more than a bunch of small capitalists squabbling over a dwindling market. But surrounding it are some large principles and historic drives:

• To save the environment, we need to conserve energy. We need to stop driving around. The ecologically safe future lies in small, dense, mixed-use communities where people do useful work near where they live.

• The single-income, two-car suburban family is the arti-

"The major decline in studio earnings nationwide over the past year has been in the area of demo recording, because musicians are making demos in their homes."

fact of an extraordinary moment in the American empire. That moment has passed.

• As their earnings decline, business owners now look to regulatory mechanisms to limit competition. Zoning laws, for better or worse, work against innovation; studios need to innovate, not regulate.

• The recording studio has become a musical instrument. Composers no longer must work acoustically and on paper, then go to a studio to record. Composition, arrangement, performance, and recording all can be done in the same place at the same time by one person using digital technology.

• In the sense that musicians are workers and commercial studios and record companies are owners, it could be said that the means of production are falling into the hands of the workers, threatening to displace a whole class of owners.

• As computer literacy spreads and equipment gets better and simpler, musicians increasingly will be able to handle their own recording. Engineers are here to stay; utter dependence on them is history.

• While in theory, technology and telecommunications bring the consumer a galaxy of choices, in practice, distribution oligopolies limit those choices. However, those oligopolies tend to neglect small markets, which creates windows of opportunity for home studio owners and artists. They can serve small markets better than the large

ES-50 synchronizer and RTS 14 rack sold separately.

companies, which have high costs and break-even points.

• The health of the semiconductor industry ensures declining prices and continuing development of digital technology. Home recording equipment will only get better and cheaper.

C ALL THEM WHAT YOU WANT: trends, megatrends, vectors, imperatives, historical drives, the utopian notions of an idle writer. The bottom line is, we are going to live differently in the future; where old and new meet, we can glimpse where we're going.

There was a time, not too long ago in the grand scheme of things, when Dad did not just go to the office and Mom did not just take care of the kids. They were both more or less in it together, along with in-laws and the grand- and great-what-have-you's. Then came the Industrial Revolution. Factories and offices took men out of the home and created not just the need for coffee in thermos bottles but, eventually, narrowly defined economic roles for Dad, Mom, and the kids. Now things are changing fast, but there is resistance to change. Laws and those who control them will be slow to adapt. And those who stand to lose economically will clammer loudly for protection.

When I was standing in the living room of that first home I visited, Lynn's husband came in from the studio to talk some business and her daughter came in from her bedroom to take the dog off me. Dad talked business with Mom while Mom talked on the phone; the daughter talked to the dog and stared at me as only teenage girls can do.

It was like a beautiful little snapshot. I'd walked into the dwelling unit of the future: Dad had finally come home from the office for good, and he and Mom were doing productive work together and maintaining a stable relationship with their children. The kid was learning about adult responsibilities firsthand by watching her parents work. That was the snapshot I wanted. That was the key to what this little war means. That told me how our society is changing.

We're coming back around again to shared child-rearing and productive home-bound economies. That's what was so beautiful about seeing that whole family standing there. In this case it was a traditional family, with husband and wife running the business, the daughter off in her room dreaming starlet dreams, and the crazy woof scaring off reporters and depositing a fine, ever-growing layer of hair and dog-sebum on the Woolite-scrubbed upholstery. But it could have been any family, any partnership, any combination. The key was the integration of life, work, and art. It was a real fine setup, a real fine way to live your life and do your art and take care of your family without bothering anybody else. And that, after all, is the American dream, isn't it? •

MIDI & COMPUTERS

Let's Get Hyper

BY MIKE COLLINS
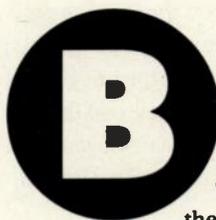
BACK IN THE DARK AGES BEFORE 1983, music products from different manufacturers remained in their own separate corners of the audio arena. The subsequent advent of MIDI allowed them all to start speaking a common language, and the MIDI community quickly expanded to include alternate controllers, signal processors, mixers, and the most powerful ally of all, the computer.

This column will focus on getting the most from these merging technologies. We'll examine the ways in which MIDI enhances the recording process, give you some useful (and fun) projects to work on, and keep you current on what's happening with musical applications for the Macintosh, Atari, PC, Next, and Amiga. This month we'll start by describing a HyperCard project that links an Apple Macintosh with a rack of MIDI-controlled signal processors to automatically log and recall any and all effects settings—at the push of a button.

• • •

It seems you can't walk into a studio these days without tripping over a Macintosh that's doing everything from sequencing to scheduling. Most Mac users are familiar with the basics of HyperCard, the "hypermedia tool kit" that comes bundled with all new Macs. One of the beauties of this popular program is that you don't need a degree in rocket science to create useful applications of your own. Those of the audio persuasion can even design customized musical tools in little more than an evening, using an add-on tool kit called HyperMIDI.

Before we get too far into this, let's take the nickel tour of HyperCard. The basic building block of HyperCard is the *stack*, a pile of similar records, such as a mailing list made of name-and-address records. On the surface, this gives HyperCard the appearance of being a fancy, computerized Rolodex. What sets it apart from the average database is its ability to link different stacks of information, as well as execute more complex tasks. A HyperCard phone/address book might handle automatic dialing via modem, play a sound effect, and animate a scanned image of the person being phoned!

When operating HyperCard in the main *browsing* level, the Mac user accomplishes all these things intuitively by pointing and clicking with the mouse.

HyperCard's *authoring* and *painting* levels allow users to create their own custom cards, which may include elements such as buttons (to initiate actions or open other cards), graphics (animated or still), and more. HyperCard's real power, however, is revealed in the *scripting* level, home of a simple yet capable programming language called *HyperTalk*. Oh no, not—(dramatic pause)—programming! Don't even *think* about being intimidated by HyperTalk, ye who have mastered every control in a studio that resembles the bridge of the Starship Enterprise.

HyperTalk scripts resemble plain English, and they let you easily attach a series of program instructions to any HyperCard elements you choose. For example, clicking on a button associated with the following HyperTalk script causes the display to flip to the next card in the stack and beep as soon as the mouse is released:

```
on mouseUp
      go to next card
      beep
end mouseUp
```

While HyperTalk offers a wide palette of versatile commands, it can't be everything to everybody. (For one thing, it doesn't know an iota about MIDI.) The language overcomes this by employing *external commands* (XCMDs) or *external functions* (XFCNs). XCMDs simply carry out some action, while XFCNs can take a reading of some external value (such as a synthesizer's MIDI volume setting), and bring that reading back into HyperCard. Both typically are written in Pascal, C, or assembly language so they can provide powerful and direct control of the computer or external devices. Many XCMDs and XFCNs are available as shareware.

Enter HyperMIDI. Nigel Redmon at EarLevel Engineering (21810 Barbara Street, Torrance, Calif., 90503; (213) 316-2939) has written HyperMIDI (Version 2.0, $195.00), a set of external commands and functions to handle MIDI operations. HyperMIDI brings a lot of MIDI power to the recording studio, as I learned when asked recently to write a custom Macintosh program for the BBC Transcription Recording Unit Studio in London. The BBC studio engineers wanted a simple, no-hassle way of dumping the current settings for all their effects devices onto a Mac disk for future retrieval. Their studio effects rack holds a Sycologic M16 MIDI Patchbay, TC Electronics 2290 digital delay, and several multi-effects devices, including a Yamaha SPX1000, two Yamaha Rev 5s, and two Lexicon PCM 70s. Everything in the rack can transmit and receive program settings via MIDI system exclusive (sys-ex) data.

> "Don't even *think* about being intimidated by HyperTalk, ye who have mastered every control in a studio that resembles the bridge of the Starship Enterprise."
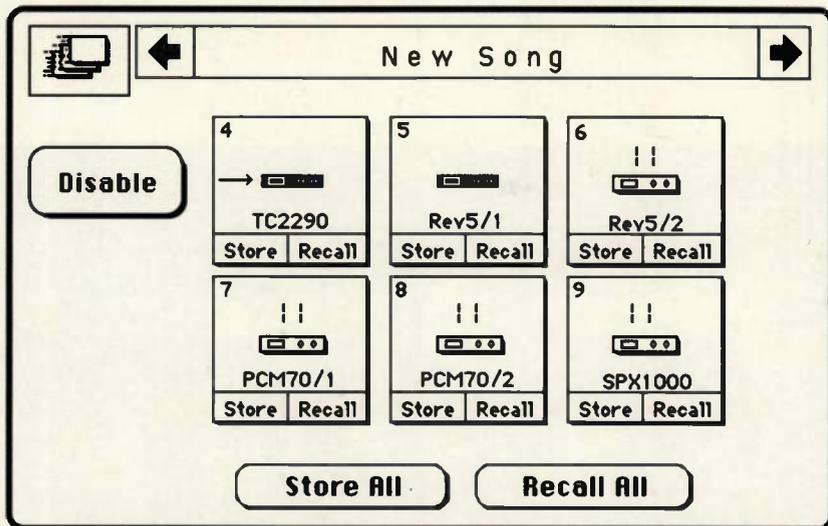
The front panel of a HyperCard/Hyper MIDI Sys Ex utility.

Many studios have the need to capture an electronic "snapshot" (on disk) of the entire MIDI effects setup for a given session so it can be recalled instantly and accurately for future sessions. There are several ways to do this. One is to use a "generic" librarian program, although these usually require significant configuration to work with a given collection of MIDI gear.

Another is to use a modern sequencer program, such as Mark of the Unicorn's Performer or Opcode's Vision; but again, this requires extra work to program bulk dump requests. The BBC wanted a *simple* program dedicated to controlling their particular selection of MIDI-controllable effects gear. My solution was to design a HyperCard/HyperMIDI application.

In the next issue, I'll give you everything you need to write your own custom HyperMIDI program for storing program data from all your MIDI devices onto a Mac disk. •

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The 2-track Open-Reel Tape Recorder

BY BRYAN LANSER

ILLUSTRATION BY MARK SWEENEY

1. Supply reel. Holds the blank.

2. Reel hold-down clamp. Attaches the reel of tape to the reel turntable.

3. Tape dancer arm (tension arm). A spring-loaded arm that guides the tape and absorbs small changes in tape tension. In a "constant tension" deck, this arm has a sensor attached that converts tape tension into a signal used to control the reel motors.

4. Tape guidance (impedance) roller. During REWIND and FAST FORWARD, it guides the tape. During PLAY and RECORD, the roller dampens out small speed variations caused by an uneven tape pack.

5. Head block assembly. Mounts the heads and guides to the top deck plate. In quality decks, this is a pre-aligned, removable assembly that allows a fast change of heads.

6. Fixed post tape guides. These precisely adjusted guides assure correct tape height across the head block.

7. Tape lifters. Fixed posts that push the tape away from the heads during FAST FORWARD and REWIND, saving the tweeters in your studio monitors from otherwise certain destruction.

8. Erase head. Imparts a high-frequency signal to the tape to randomly orient (erase) the magnetic particles on the tape.

9. Flutter roller. A rotating cylinder that prevents the tape from vibrating.

10. Record head. Records the tape by organizing its magnetic oxide into a pattern analogous to the input signal fed to the deck. During SEL REP operation—when a new track is recorded in sync with a previously recorded track—the record head is used to play back the prerecorded track.

11. Head shield. A retractable metal shield that covers the play and record heads, and prevents external magnetic fields from interfering with the record or playback processes.

12. Play head. Converts the varying magnetic patterns on the tape into electricity, which is then amplified and equalized by the playback electronics (electronics not shown).

13. Capstan shaft and motor. Drives the tape at a precisely controlled speed during PLAY or RECORD, and ensures stable pitch.

14. Pinch roller. A soft rubber roller that pushes the tape against the capstan shaft during PLAY or RECORD.

15. Pinch roller solenoid. Controls the pinch roller. Some higher-priced recorders are of a pinch-rollerless design.

16. Tachometer roller. Generates a pulse signal that is used by the tape location circuitry to keep track of the tape position, and guides the tape.

17. Take-up dancer arm (tension arm). Same as supply side—with the exception that this usually has a switch attached to it, which turns on the capstan motor or kills the motor power at the end of the tape reel or in the event of tape breakage. Some decks use a photo-switch to detect the tape, instead of a switch attached to the arm.

It's all well and fine to read articles about how gear works. But sometimes a picture is a far more eloquent teacher.

Each issue, "How It Works" will explode microphones, cut away mixing consoles, and take apart digital audio microprocessors. It's our hope to make this feature both informative and entertaining for you, whether you're a seasoned audio professional or a beginning recordist.

This time we've cooked up our own version of a generic 2-track analog open-reel tape recorder. Of course, not every component or function will apply to every 2-track out there, but we've touched upon the important ones.

If there's something you'd like to see in a future "How It Works," please send your suggestions to How It Works, c/o EQ magazine, 20085 Stevens Creek, Cupertino, CA 95104.
—Editor

18. Take-up reel. Stores the tape as it is fed by the supply reel.

19. Reel turntable. The surface upon which a reel of tape is mounted. It's attached directly to the reel motor.

20. Brake band. Holds the reels in place during STOP mode and slows down the reels when bringing the tape to a stop.

21. Brake release solenoid. Disengages the brakes and allows movement of the reels when the transport is running.

22. Brake drum. Where the brake band makes contact to slow the tape.

23. Reel motor. Drives the reels of tape in FAST FORWARD and REWIND. It also provides take-up or hold-back tension during PLAY and RECORD.

24. Main power switch. Turns the whole mess on.

25. Varispeed adjustment control and display. Speeds up or slows down the capstan motor from its normal operation speed. Our unit displays tape speed in inches per second or as a percentage of normal speed.

26. Speed select switch. Selects the operating speed of the transport. Most open-reel decks offer two or three speed settings. Common 2-track speeds include 3-3/4, 7-1/2, 15, and 30 ips.

27. Cue lever (lifter defeat). Places the tape against the heads as long as it is engaged, allowing the recorded signal to be heard during FAST FORWARD or REWIND, or when the reels are rocked back and forth during editing. Most lock in position.

28. Location display (program counter). Displays the current location of the tape; in this case, in hours, minutes and seconds.

29. Autolocator. Allows the user to pre-program different cue points, allowing different points on the tape to be located. Most current 2-tracks store two or three cue points, plus ZERO.

30. Dump edit button. Disengages the take-up

motor to allow tape to be dumped from the supply reel during editing.

31. Record. Causes the deck to enter the record mode when pressed simultaneously with the PLAY button or when pressed alone when the deck is in PLAY.

32. Play. Moves the tape forward at the selected speed and reproduces the signals on tape.

33. Stop. Stops the tape when the transport is in PLAY, RECORD, FAST FORWARD, or REWIND.

34. Rewind. Winds the tape onto the supply reel while causing the tape lifters to push the tape away from the heads.

35. Fast forward. Winds the tape onto the take-up reel while causing the tape lifters to push the tape away from the heads.

36. Splicing Block. Holds the tape in place for cutting and provides an angular path for the razor blade.

37. VU meter. Displays the level of the signal being fed to the machine or reproduced off tape.

38. Record ready switch. Prevents entry into RECORD when in SAFE position.

39. Channel status switch. Selects between INPUT, SEL REP (see item 10), and REPRODUCE modes of operation.

40. Record and reproduce level controls. Adjust the level of the input and output signals of the deck.

41. SRL switching. Selects the precalibrated Standard Reference Level for use in a fixed-level application. In professional settings, this usually is a +4dBm level; the personal studio setting usually is -10dBV.

42. Audio calibration controls. Calibrates all levels in the deck such as gain, equalization, and bias.

43. Monitor level control and output. Adjusts the level of the headphone output jack.

44. Power supply. Provides the necessary voltages to the transport, audio, and control electronics. •

STUDIO CLINIC

Tape Path Maintenance

BY RICHIE MOORE, Ph.D.

MOST RECORDING EQUIPMENT built these days is designed to function with minimal or no maintenance. But notice the word "function." There can be a huge difference in performance between a device that's simply left to do the job, and one that is well-maintained and optimized.

For instance, every analog tape recorder is calibrated at the factory for at least reasonably good performance. By the time it's shipped to its destination—and this applies to everything from a 24-track on down to a cassette deck—its calibration may be out of whack. Add to this a month or two of operation and chances are, you have a recorder that is falling well short of its optimum performance in terms of frequency response, signal-to-noise, and other factors. So you clean your heads. Great. And most likely you demagnetize. Wonderful. But what can be done beyond the obvious?

In fact, you can do a lot to improve the performance of individual components, your entire system, and your studio. It usually doesn't take a lot of test gear to maintain your studio, and the results are well worth the bit of work required.

Let's start off with a look at the tape recorder and its tape path. This is basic stuff, maintenance you certainly should be doing if you own a recorder. However, as we'll learn, there are right and wrong ways of performing even the most simple procedures.

• • •

The tape path of a magnetic tape recorder is its weakest link. Regardless of the machine—digital open-reel or cassette—the tape path requires constant vigilance. The main components of the tape path are the *reel turntables*, *tape arms* and *guide rollers*, *guides*, *tape lifters*, *flutter roller*, *pinch roller*, *capstan*, and *headstack*. All of these must work in harmony to maintain quality audio. [*Ed. Note: Turn to "How It Works" on page 50 for a diagram of these components.*]

Studio Clinic will deal with a wide range of topics, from consoles to room tuning. I'll try to present information that is useful to both commercial and personal studios. If there are certain issues you'd like me to tackle, please drop me a line, c/o EQ (20085 Stevens Creek, Cupertino, CA 95014). Your suggestions may make their way into a future column.

Clean Sweep. It is extremely important to clean your recorder's tape path regularly. Oxide (the magnetic surface of the tape), lubricant, and miscellaneous dirt shed from the tape and accumulate on these parts. In no time at all, they build up and cause degradation in audio performance, slippage, and undue wear on the tape. Rule #1: Clean the tape path at least once every few hours that you run the tape recorder. Some diligent engineers clean it every hour.

Never use any abrasive material to clean the tape path. Many engineers use a cotton swab, such as a Q-Tip, to clean heads. Avoid using a cheap swab that might leave lots of fiber behind. Also, choose a swab that has a wooden or paper shaft: Plastic-shaft swabs can "melt" when used with certain cleaners. Better than a Q-Tip, but a bit more costly, are a lint-free cloth and a sponge video-head swab.

Which head cleaner solution is best? Some engineers prefer Freon TF, since it evaporates quickly and leaves no residue. Unfortunately, this is the same stuff that's currently destroying our planet's ozone layer. Somewhat more ecologically sound are commercially available head-cleaning solutions, which usually consist of CFCs. [*Ed. Note: Turn to "Studio Solutions" on page 64 for a cleaning supplement.*] Pure (97-99 percent) isopropyl alcohol also works quite well, though some people feel its slight water content can cause some rusting in the long term. Others prefer methanol, which contains no water or impurities and dries faster. Whichever you choose, use it with caution; many of these various formulae are hazardous to your health.

As for cleaning the rubber components (including the rollers and some tape guides), use *only* pure alcohol and a lint-free cloth. This is to avoid embedding any dust or lint particles on the surface of these rollers. Some manufacturers recommend a special rubber cleaner (TEAC makes such a product), which is designed to prevent the rubber from drying and cracking. This stuff probably works fine, but I'm not sure it's necessary

The Basics. The actual cleaning process is straightforward, and is basically the same for a cassette or open-reel transport. The idea is to clean every component that the tape comes in contact with. Dip the swab (or cloth) in the cleaning solution, and get scrubbing—gently. When the swab begins to look slightly soiled, grab a new one. Assuming you don't use too much fluid, there's no need to dry off the surface with another swab; the solution should evaporate in a few moments.

Follow the entire tape path, starting at the left with the tape lifters (on open-reel decks only), then move on to the guides and headstack. Use the cloth when you get to the rubber pinch roller. Be sure to scrub under the edges of all the guides for particles that may be trapped in these areas. You can't always see the oxide build-up here, but it really accumulates fast.

It's best to clean the capstan (the metal shaft that rests against

> The tape path of a magnetic tape recorder is its weakest link. Here's how to keep it strong by keeping it clean.

BASIC TRACKS

Preparation Is The Key

BY TOM LUBIN

RECORDING IS ABOUT creativity. It's also about technology and technique. As one becomes more fluent with the process of recording, the technology becomes less intimidating, the techniques become more natural, and the creativity begins to flow.

Certainly, recording is a fun process if you remember that your goal is creativity. But fluency in the recording studio doesn't come easily. It takes work, and a willingness to learn. It's a special challenge if you're recording on your own. And sometimes even the most advanced of us need a brush-up course once in a while. This is where "Basic Tracks" comes in.

In each issue we'll deal with fundamental concepts of recording. If you're a serious, motivated amateur—or a well-seasoned pro looking to review some fundamental concepts—it's my hope that this column will serve you well. We'll look at all angles of recording, with a strong emphasis on recording in the personal studio. We'll also, from time to time, deal with concepts more oriented towards recording in the commercial studio—such as in this first column.

• • •

Seven Ridiculous Things People Try To Do In A Recording Session.
1) Invent equipment.
2) Form bands.
3) Write songs.
4) Negotiate contracts with managers.
5) Learn new instruments.
6) Fit a 30-voice choir into a 4-person studio.
7) Bring in three singers who never worked together before and expect them to sing backgrounds to five songs in three hours.

Actually, I could make this list a lot longer. But the point is this: These experiments in the studio simply take much more time to realize than most people can afford. It is *crazy* to rent a gizmo that nobody knows how to use and expect to get what you want out of it quickly. It makes *no* sense to hire hot horn players and bring them in without charting and rehearsing your horn arrangements in advance. In short, it's a waste of money to walk into a recording session without being properly

prepared. Let's take a look at how thinking ahead can turn your time in the studio into a wise investment.

Most people who record in small personal studios do so on their own, or with one or two other people. Few of their recording sessions ever use a human source of rhythm—rather, the ubiquitous drum machine and sequencer crank it out. Many of today's commercial sessions also use these electronic rhythm machines. Today, in fact, many people don't play music *together* anymore: They play it solo.

But what if you're preparing to take a *band* into a commercial studio—either as a musician, an engineer, or a producer? The process of recording a band is completely different from working solo. Furthermore, most commercial studios present a scene unlike what you're used to in your personal studio.

Preproduction Makes Perfect. Before you step into a commercial studio, remember the proverbial running clock: With studio rental costs ranging from $20 to over $200 per hour, it's the wrong place to dilly-dally.

For this reason, *preproduction* is essential. The days of going in and playing a 16-bar groove, then trying to write lyrics that make sense, just don't exist anymore. Now most songs have to be developed and crafted before the tape rolls.

Preproduction often consists of lots of rough demos on a 4-track, or an ordinary cassette machine, or even on a sequencer and drum machine, to sketch things out. It is equally important to allow preproduction time for vocals and harmonies.

Whatever isn't preproduced before the session often ends up in the frustrating realm of trial and error. (Maybe magic will save the day, but usually not.) Furthermore, learning the lyrics and melody on the same day you're recording the song won't produce a convincing performance. Songs recorded this way often have elements of hesitation.

Most importantly, the preproduction process gives you the chance to experiment with different approaches, to find the arrangements and sounds that work best.

Make Your Bed. "Bed tracks," or basic tracks, form the foundation of a song. They're usually the first tracks recorded on a blank roll of tape. Typically, they consist of drums, bass, rhythm guitar, and perhaps some keyboard and a "scratch" vocal. They also may include a click or metronome track (for timing), and a SMPTE time code track, to synchronize a drum machine or sequencer (or video) to tape. They're the basics. Anything fancy comes later.

Traditionally, the bed tracks for at least one song—and sometimes several songs—are recorded during a single session. Many of the best-sounding bed tracks are recorded live—that is, with the musicians

> "It makes no sense to hire hot horn players and bring them in without charting your arrangements."
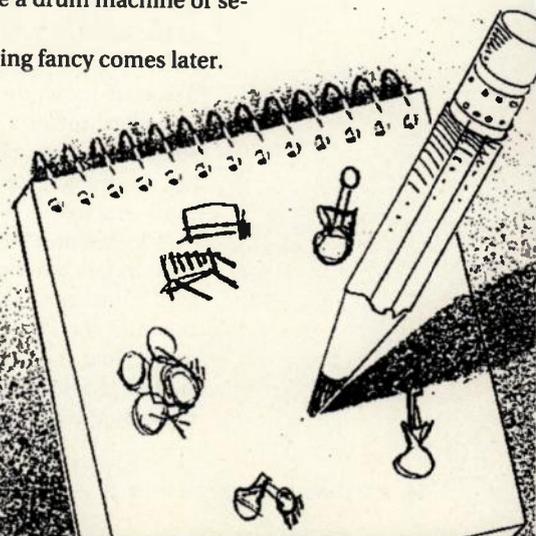
FILM & VIDEO SOUND

The Challenges Of Post-Production

BY LARRY BLAKE
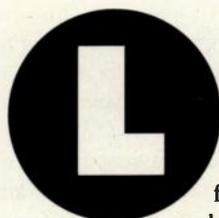
LIVING IN THE WORLD of post-production sound involves fighting a constant battle against The Odds. Whether you're recording, editing, or mixing sound for television or feature films, you're always faced with logistical problems: How can you complete a project on time and on budget, yet end up with something you'll be able to look at five years down the road?

But logistics are only the tip of the iceberg. There's the frustration of the process itself. Time is spent—and wasted—on sync problems, or making dozens of changes to your carefully assembled soundtrack because the picture has been re-edited. Then there's the whole issue of personalities: You have to please picture editors, directors, and associate producers, each of whom will be only too glad to offer his or her opinions.

In this column I'll be giving you some tips that will save you time and money. At least they'll save you a little grief. Before we get into problem-solving, let's examine the various aspects of post-production sound work and define a few terms.

Dialog Editing. The first step of this process really hasn't changed much in 60 years. The sound editors receive a *worktrack* which contains the original production sound that was recorded during filming. They also receive a *dupe* (usually in black-and-white) of the edited picture; this is known as a *workprint.* Although listening to the worktrack in sync with the workprint will give the editor a good idea of what the film ultimately will be like, even under the best of circumstances there's still a great deal of work to be done. Creating a feature film soundtrack typically involves mixing ten tracks of dialog, dozens of effects tracks, and two or three pre-mixed music tracks.

The first and foremost task facing the sound editors is to smooth out the production dialog track. The dialog track from each individual camera angle is first split into separate tracks. Each angle has its own background sound, and the sound often differs from one angle to another. To help smooth out the changes in background noise as the picture cuts from one angle to another, some *room-tone handles* (the ambient sound of the room when no one is speaking) are added to the head and tail of each angle. After all the editing, the re-recording mixer then crossfades these handles to create seamless sound transitions from one shot to the other. Room tone also is used to mask undesirable noises, such as the director's instructions or camera dolly creaks.

Replacing Dialog. When the production dialog is unusable, or the director wants a better reading, the dialog editor tries to substitute ("cheat") an alternate take for the one that is to be replaced. For instance, although the picture may come from the first take of a scene, portions of the sound from take three may be used. If this process is unsuccessful, the only other recourse is to re-record the line(s) in the studio. This process, known as *post-synchronization, looping,* or *ADR* (automated dialog replacement), requires that the dialog—often hundreds of lines—be recorded and synced, syllable by syllable. This is done by having the actors say the desired lines while they watch the accompanying scene.

The final dialog tracks sent to the dubbing stage sometimes consist of a staggeringly complex quilt of original production tracks, cheated production tracks, and ADR tracks. Individual sentences actually may consist of dialog generated in all three ways.

Sound Effects. Sound effects generally can be grouped into three categories: backgrounds (ambience), "hard" effects, and Foley (named after the man who pioneered the process.) Backgrounds, such as wind or birds, not only help set the time, place, and mood for a particular scene, but also smooth out inconsistencies in the production track. They are also helpful for the "minus-dialog" music and sound effects tracks used when a film is dubbed into a foreign language.

Hard effects cover sync-specific actions on the screen, such as a door closing, an explosion, or a rocket ship taking off. These sounds are pulled from a sound effects library, recorded specifically for the film, or created with synthesizers and digital samplers. When a hard effect is performed live in sync with the picture, it's called Foley. Footsteps and door creaks are common Foley effects. Sometimes Foley is used to create a "signature" effect, such as alien footsteps.

Putting It All Together. Soundtracks can be assembled using analog or digital multitrack tape, a digital audio workstation, or traditional 35mm mag film. Frequently, all three media come into play during the course of a soundtrack's development. I prefer to work with multitrack tape that is locked to video picture via SMPTE time code. Mag film, however, is still very popular, and the multitrack tape versus mag film debate continues to rage. Next time, I'll tackle this issue head on. •

Some of you may recall my writings in Recording Engineer/Producer. *One of the joys of writing a column is hearing feedback from readers, so feel free to contact me care of EQ, 20085 Stevens Creek Blvd., Cupertino, CA 95014.*

> "A film soundtrack typically involves mixing ten tracks of dialog, dozens of effects tracks, and pre-mixed music tracks."

THE BIG PICTURE

Audio Meets Video

BY ROB WAIT
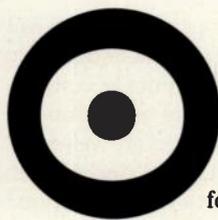
O N THE WAY HOME FROM the pet store, where you just picked up a couple of Martian mice to feed to your Venusian saber-toothed kittens, you're struck with a great idea for a song. You *hear* every note, *see* every image for the video.

Luckily you just happen to have your pocket audio/visual recorder in, you guessed it, your pocket. With hardly a swerve of the wheel, you deftly put on the recorder's headset that doubles as a thought interface for most current computer-type devices. You rethink all the parts of the song as the recorder encodes its molecular disk to reflect exactly what you're thinking, with 99.9999 percent accuracy. This is mainly an audio pass, but a few stray visual images also sneak in. You go back and combine these images with memories of your last trip to the outer solar system, plus a few thoughts of that gorgeous cashier who gave you a second glance at the Gourmet Food Pill Store. (It's *still* pop music, after all.)

Loud squeaking from the back seat suddenly breaks your concentration. You realize with horror that the pet shop must have mixed some Rapid Growth Plutonian Attack Mice in with the Martian mice. You quickly hit playback. Your new song and video blast over the speakers and onto the windshield screen. The mice like it. They decide not to eat you after all, and simply ask to be let out at the corner.

This image from the future (save for the mice, perhaps) isn't too hard to imagine if you've been keeping up with the latest audio/visual technological advances.

In both audio and visual arts, we're progressing rapidly towards smaller, faster, easier-to-use, less expensive equipment that ties together so simply that, even with little technical knowledge, creative types can express themselves with a relatively low outlay of capital, brain cells, and equipment-hauling back strain.

Whatever your involvement in the audio business, sooner or later you'll run into a situation that calls for a basic knowledge of the workings of film and video. The goal of "The Big Picture" is to give you that knowledge, and to venture into more complex concepts surrounding the fusion of aural and visual technology. We intend to cover the entire range of possible audio/visual situations, including the basics of how to sync a MIDI system to video for scoring, how to shoot a garage video on a shoestring budget, and how to step in front of a multimillion dollar, random-access, film/video/music editing and mixing system and say something more intelligent than, "So where's the gas pedal?"

• • •

Sometime around the dawn of time, one guy decided he could record sounds for posterity on a phonograph of sorts, and another guy decided he could do the same with visuals on moving-picture film. Then a couple more ambitious trendsetters figured out both inventions became more exciting if you put them together. The film industry boomed, and engineers endeavored to make movies look and sound big and impressive. Bear in mind that while sound and picture were working in tandem, they basically were tied together by a crude mechanical method of synchronization. Sound engineers concentrated on ways to improve sound equipment, while film stock companies worked on perfecting the visual image. As long as sound and picture received separate treatment, the industry stood little chance of inspiring the future suggested in our opening paragraphs.

Then came Television. Both picture and sound came in the form of electrical signals that could be transmitted to, and received by, a single box. Even though the picture and sound still were dissimilar, they at least *appeared* more unified and easier to deal with. (Simplistic, but we're only after general concepts here.)

After the invention of television, and then the invention of the computer, came digital recording. Finally this gave us a format that would allow techniques learned in audio engineering to apply to video, and vice versa.

Since the early 1980s, digital synthesis, MIDI, and sampling have permeated the music world, enabling us to gain access to more sounds and options simultaneously than we ever dreamt of.

Similar excitement has infused the land of film and video. Cameras are better and smaller, film editing systems cross the line between video editing systems and random-access computer technology, and music television and home video have made the viewing public more visually literate and avid for better, more adventurous visual products. Consumers, by experimenting with video production techniques, are creating a demand for technological advances that force manufacturers—if they hope to stay in the market—to produce equipment with more features for less money. All in all, business is better than ever.

This has been a brief zoom-in on the independent development of film, video, and audio technologies. We've only recently started to approach the point where all three will merge into some shape that will be touted as the entertainment of the future.

As things develop, "The Big Picture" will provide you with previews. •

> **"Sooner or later you'll run into a situation that calls for a basic knowledge of the workings of film and video."**

the pinch roller when it's running) while it's moving. Make sure there's no tape on the machine, then put it into PLAY. On open-reel decks such as those by Otari, Tascam, and Scully, you may have to switch on the capstan motor by holding up a tape arm. On Studer and other decks that use a light sensor to detect the absence of tape (and thus switch off the capstan at the end of a reel), you may have to place a bit of paper between the light bulb and the sensor in order to turn on the capstan motor. In any event, once it's running, place a swab against the capstan, and give it a good cleaning.

Demag That Deck! Cleaning the tape path is the most fundamental bit of routine maintenance. Close on its heels, though, is *demagnetization*. As you may know, magnetic tape can magnetize metal components it encounters. Once it magnetizes a recorder's heads, the most readily audible result is a loss of high frequencies. If it has been some time since you last demagged, you're robbing yourself of good high-end response. At the least, demag your deck after every ten hours of use.

However, demag with *extreme caution!*

Keep all tapes, floppy disks, and computers a good eight feet away from you during this process. Make sure the power at the tape machine is OFF. Use only a demagnetizer with a high flux density, such as the R.B. Annis HandiMag; inexpensive "hi-fi" types can do more harm than good. Turn on the demagnetizer a full arm's-length from the machine, and bring it slowly towards the tape path.

Work left to right on each item in the tape path. Start by bringing the tip to within 1/8 inch of an individual tape guide, lifter, roller, or head. As you get close, slowly move the tip of the demagnetizer up and down (or in slow, very tight circles). Never turn the demagnetizer off when you're near the tape deck—otherwise you may deliver a strong, unwanted charge to the deck. When you're finished, slowly move the demagnetizer at least three feet away.

Repeat these steps for each guide, head, and roller in the tape path. For rollers, be sure to keep moving the tip up and down (or in circles) along each roller so the entire surface is exposed to the demagnetizing field. Finally, make sure the demagnetizer is at least three feet away from the deck before you turn it off.

• • •

All tape path maintenance procedures require a little time and effort. But the results, on both the sonic and operational levels, can be astounding. Next time we'll take a look at the height of the tape path and some important issues concerning head alignment. Until then, remember that preventive maintenance is the best maintenance. •

playing together. The groove created by a good drummer and bass player working together is often much better than two tracks recorded separately.

I have worked with producers who will record one song's basics, and continue to do overdubs for that song until its completion, before recording the basics for any other songs. If the musicians are well organized and well rehearsed, it's possible to complete the bed tracks for two or three songs in just four to five hours. Studio musicians in Nashville, for instance, are noted for their ability to lay down three cuts or more in a standard four-hour session.

Next time we'll take a look at the process of recording bed tracks, and how to set up the studio for a live band recording. •

Secrets Of Vocal Sampling

PROFESSIONAL TIPS TO OVERCOME THE FEAR OF FLYING IN VOCALS

To err is human. To err repeatedly is to be a human in a recording studio. To put up with repeatedly erring humans is to be a recording engineer. And to put erring humans, an engineer, a perfectionist producer, and a clock-watching client in the same room together is to re-enact an episode of *Nightmare On Elm Street.*

You've probably experienced at least one (if not all) of the following studio nightmares: The lead vocalist can't hit the high note (of course, this isn't discovered until after the entire track has been cut); the background singers don't blend; there are popping P's and other plosives in the vocal track; there are bad notes that are too tight to punch-in or -out on; there aren't enough tracks (there never are); there aren't enough outboard effects devices (ditto); there isn't enough time, money, patience, personnel, talent—you name it. Thank goodness for digital technology. With its help, we can keep your nightmares at bay. →

BY MICHAEL MARANS

ILLUSTRATION: SAROYAN HUMPHREY

Samplers and digital recording and editing systems afford us virtually unlimited ways to manipulate sounds and performances. With them, we can cut, copy, erase, replace, and process recorded data to our hearts' content. And we can do all these things without fear of losing our original recording. A typical dance mix, for example, once involved cutting and splicing hundreds of pieces of audio tape—some of which were only inches in length. These days, the same mix (in fact, hundreds of mixes) can be created easily on a digital audio workstation. All the while, the original recording remains safely tucked away on a hard disk.

Many of the same techniques made possible by digital workstations also can be performed on the average digital sampler, especially when it comes to vocals. You can create a lead vocal track out of words and phrases taken from multiple takes, pitch-shift a vocal to correct a bad note, and raise or lower an entire vocal track to place it in a new key. You can create ambience, delay, echo, chorus, and stereo imaging and panning effects without using outboard gear. You can use the "perfect" vocal take, such as one sung in a chorus, each time that part appears in the song. Need a bit more convincing?

Since samplers are MIDI devices, vocal performances can be recorded into a MIDI sequencer before you go into the studio, which will save you time and money. (We knew you'd like the part about saving money.)

To Tape or to Sampler?

There are two basic approaches to sampling vocal parts. The easiest method is to record all the tracks of a given part on multitrack tape, process and blend the parts as desired, then sample the final mix. You'll also be working from tape if you're fixing a bad part.

Recording the vocal parts directly into a sampler provides a number of advantages, including better sound quality (no tape hiss) and the ability to process each part independently. Also, you can save time as you'll only need to record one good take for each part; you can use the same take each time that part appears in the song.

The First Session

Let's presume that you've decided to record your vocals directly into the sampler. The setup would be identical to a standard vocal session except the vocal signal (from the mixer) would be routed to the sampler's input, rather than to a tape track. All EQ settings and level blends should be accomplished at this time. You also may want to add whatever effects processing you plan to use. When you're happy with the overall sound of your vocals, it's time to set up your sampler's record parameters.

Determine the total length of all the samples that you wish to take. (Remember that you can use the same group of samples each time they need to appear in the song.) Choose a sample rate that provides enough sampling time to record all the parts. The preferred rate is 44.1kHz, but you may be able to use a rate as low as 30kHz and still maintain an acceptable level of fidelity. Next, set the record level, select the manual sample mode (as opposed to threshold or "trigger" mode) to insure capturing the complete attack portion of the first syllable, and record all the vocals.

A word of caution: When a sample is recorded, it re-

Basic Requirements

RECORDING VOCALS into a sampler is much like recording them to tape. You can easily adapt your recording equipment and techniques for use with a sampler. Here are things to consider:

Choosing a sampler. Since the human voice doesn't contain a great deal of high-frequency information, usually it can be recorded satisfactorily on a mid-priced 12-bit device (such as an Akai S900, Ensoniq EPS, or E-mu Emax). For **professional** applications, however, samplers with 16-bit linear resolution like Roland's S-770 and Akai's S1000 offer considerably better sound quality and more sophisticated editing/processing utilities. High-end machines also may contain as much as eight megabytes of memory, which translates into over 90 seconds of recording time (48 seconds in stereo) at a 44.1kHz sample rate. Hard disk-based recording systems also offer considerably longer record times (typically, one minute for every ten megabytes of disk space). A good 16-bit sampler and a disk-based system both can produce CD-quality audio. This article uses the word "sampler" to refer to a sampler or disk-based recording device.

A sampler's processing utilities are important for cleaning up bad parts, and for creating effects. These utilities include truncation, crossfade looping, digital splice/combine, filter, filter and amplifier envelopes, and non-transpose key mapping. Less essential but desirable utilities include graphic and/or parametric EQ, scrub wheel, digital pitch-shift, sample-rate conversion, and time-compression/expansion. You'll find most of these in computer-based waveform editing software, even if your sampler has none built in.

Signal processing devices. Whether you process your vocal parts before they are laid to tape, or you cut them dry, use the same techniques when you sample them. If you're sampling the vocals from tape, use an exciter to help maintain the vocal's sizzle. This is especially important if you use a 12-bit sampler, as the frequency response of these instruments usually drops off sharply at around 11kHz. A compressor/limiter can help smooth out the peaks and valleys in the vocal's amplitude, resulting in a better signal-to-noise ratio.

Microphones. Choose these as if you were recording to tape. A large-diaphragm condenser (such as an AKG C-414 or Neumann U87) will give the best results; if you're on a tighter budget, a quality condenser such as a Shure SM87 or Toa K-1 should do the trick.

—*Michael Marans*

sides *only* in the instrument's internal memory, which is volatile. If the power fails, or if you turn off the sampler, all the samples in memory will be lost. So it's a good idea to save the recorded vocals periodically to disk. That way you'll always be sure your work won't be lost if an accident occurs.

When manual sample mode is used, there may be some dead air at the beginning of each sample. This dead air must be truncated (cut) so the sample will play back immediately when a key is pressed.

If you have a visual editing system, you can easily determine what part of the waveform is dead air and what part is vocal (Fig. 1). The dead air will appear as a relatively straight line at or near the ZERO AMPLITUDE axis. This line will stretch from the beginning of the sample up to the point where the waveform begins to get dense—the actual vocal signal.

If you are truncating the sample using your sampler, here's a trick to help you perform quick and accurate truncations:

1) Enter the sampler's truncation menu and set the SAMPLE END point to zero, or its shortest possible length.

2) Now play a key; you'll hear only the very beginning of the sample, a.k.a. dead air. Gradually increase the END POINT value and repeatedly press the key. Continue this process until you hear a click at the end of the sample. This click is the sound of a single sample being played; in this case, the start of the vocal data.

3) While repeatedly playing a key, reduce the END POINT value one sample at a time until you can no longer hear the click. You eventually will reach a point where a given end number plays silently, while the next higher number produces a click.

4) Make a note of the end number that plays without making a click. Enter this number for the SAMPLE START value, and return the sample end to its maximum

Create a lead vocal track out of words and phrases taken from multiple takes. Pitch-shift a vocal to correct a bad note. Raise or lower an entire vocal track to place it in a new key. Here's how.

value.

5) Check to make sure that the sample plays properly and that it triggers immediately when the key is pressed.

6) When you're satisfied that the truncation values are correctly set, execute the TRUNCATION function to remove the dead air before the new start point from memory. Once again, be sure to save your work to disk.

Performing the Vocals

If you have recorded individual harmony parts that will play back together, assign each of the parts to adjacent keys (for example, *C1, D1, E1*) so you can play them easily. Use white keys only; it's easier to remember which sample is assigned to which key, and they're easier to play simultaneously. When recording group parts, assign each new sample to several adjacent keys, so you can produce doubling and chorusing effects by playing multiple keys.

Let's assume you have a four-line chorus such as, "Ooh baby, baby, I love you, Ooh baby, baby, You're so cool." (Yeah, we know. Deep and meaningful.) Assign the samples as shown in Fig. 2. Each vocal line spans a four-key range and is in non-transpose mode (each key plays the sample at its original pitch). Assign a moderate amount of velocity to VCA sensitivity across the entire note range. This allows the volume of each sample to be con-

trolled by your touch.

Notice that only the first, second, and fourth lines of our chorus have been assigned. In our example, the first and third lines use the same lyrics and melody, so we can use the same sample for both lines. Additionally, each time the chorus appears in the song, the background vocals are the same. This allows us to use the same three samples for every chorus (even for the 37 repetitions that occur in the fadeout). This saves sample memory—not to mention the time it would take to record new vocals for each and every chorus.

Real-Time Effects

The non-transposed key mapping just described allows us to create the following effects:

• *Flanging.* Simultaneously depressing several keys will produce flanging effects. The depth and tonality of the flanging is determined by the amount of delay between the individual note-on events.

• *Room ambience.* Play four keys in a row as though you were quickly drumming your fingers on a tabletop. The degree of ambience is determined by the speed of the drumming.

• *Slapback echo.* Play each key successively at quarter-, eighth-, or sixteenth-note intervals, depending on the degree of slap desired. Use velocity to control the volume of each phrase. The first note should be the loudest, and each additional note softer than the one previously played.

If you are sequencing your vocal parts rather than playing them live, you can create delays (or eliminate them) by shifting the start time of a sample's NOTE-ON by a few clock pulses. You also can enter MIDI VELOCITY values for each note, which will enable you to control their respective volumes more accurately.

Programmed Effects

You can create a number of dynamic effects using the analog and digital editing parameters found on most samplers. To create these effects, each of the samples in a group is processed differently.

• *Chorusing.* Detune two of the four samples in each group by ± 9 to 12 cents from original pitch. When all four samples are played simultaneously, the vocals will sound chorused. To create a dynamic

VOCAL SAMPLING

chorus, assign a small amount of LFO modulation to the pitch of one or more of the samples.

• *Digital echo.* Determine which part of the vocal phrase you wish to repeat, such as the first "baby" from the phrase "Ooh baby, baby." On your sampler, define LOOP START and END points that result in the word "baby" repeating for as long as the key is depressed. Next, turn off the RELEASE LOOP so the sample plays to its end once the key is released. A VCA envelope can be used to control the number of repeats (see Fig. 3).

• *Stereo imaging.* This technique is similar to using a digital delay unit to split and double a sound, allowing you to create the appearance of full stereo using mono samples. Assign two of the four samples in a group to the left side of the stereo spectrum and the other two to the right side. Next, delay the samples on the right by 20 to 50 milliseconds. If your sampler does not have a delay function, you can splice dead air to the beginning of the sample you wish to delay. If you are sequencing, you can delay notes by offsetting them by a few clock pulses. Lower the volume of the samples on the left by a slight amount (usually a dB or two is enough). Play all four keys simultaneously. To create dynamic panning effects, assign an LFO, mod wheel, or other MIDI controller to the panning function of your sampler.

The techniques described here create just a few of the many vocal effects that may be developed on a sampler. You can use filter sweeps to turn an ordinary vocal into an unusual sound effect. You can pitch-shift a phrase down an octave and play it against the originally pitched sample. If backward masking is your thing, digitally reversing a sample is as easy as pushing a button. The key word here is *experimentation.* With a little luck, those "insignificant" background parts could become your song's best hook.

Fixing Problems

In the introduction we mentioned a number of problems that would be difficult to remedy using conventional techniques. Here are some digital fixes to a few unpleasant situations you're likely to encounter during a typical vocal session.
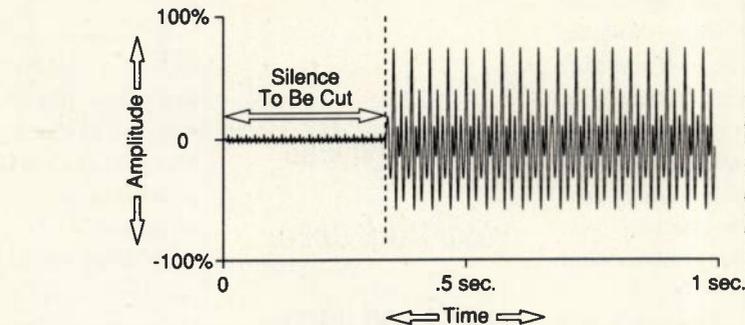
Fig. 1. Truncating initial silence insures immediate response on triggering.
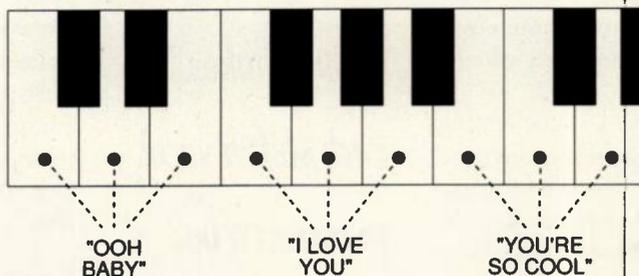
Fig. 2. Mapping the same sample to consecutive key enables effects without external processors.
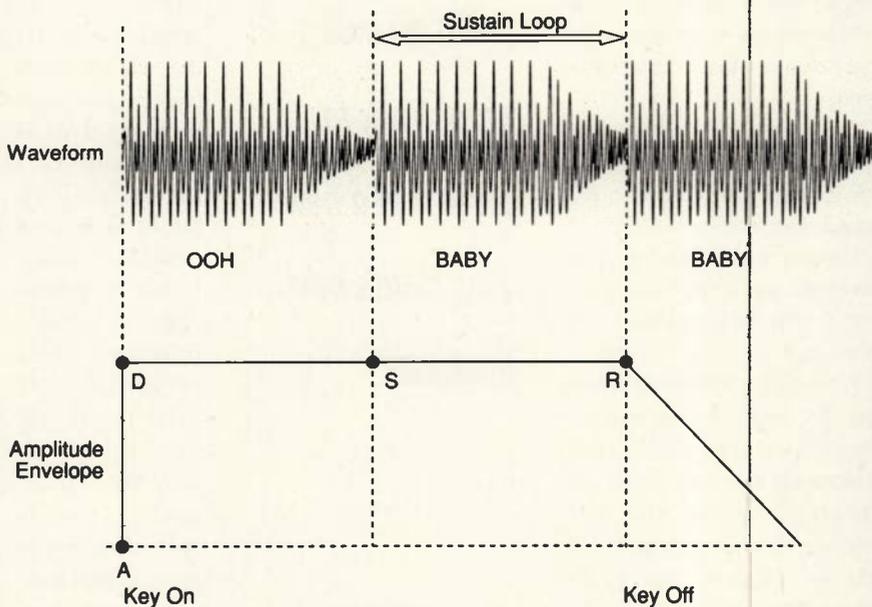
Fig. 3. Sustain loops can perform automatic repetition.

High hopes, limited range. Sometimes the lead vocalist just can't hit that high note as cleanly as you'd like. Fortunately, you can expand his or her range simply by using the sampler's pitch-shift capabilities.

To correct notes that are flat by a consistent amount, sample the phrase that contains the bad note from tape. Cut the bad note out and copy it to another sample location. Digitally pitch-shift the note by the amount necessary to bring it up to the correct pitch, and splice the pitch-shifted note back into the original data.

Use the pitch-bend wheel to correct notes that are off by different amounts over time. Sample the entire phrase, then practice playing back the sample while

adjusting the pitch with the wheel. Once you have perfected the move of the wheel that provides the proper pitch correction, record your "performance" into your MIDI sequencer. Sync the sequencer to the tape, and make any adjustments necessary for the phrase to begin playing back at the proper point. Now simply punch in the phrase over the original on tape.

Pitch-shifting on a sampler is not without its drawbacks. The biggest of these is that pitch-shifting usually results in "munchkinization"—the same effect used to create the voices of Alvin and the Chipmunks. Recent advances in digital signal processing (DSP) technology, however, have resulted in algorithms that allow pitch to be shifted independent of time. This lets you alter the pitch of words, phrases, or even entire vocal tracks without encountering the problems normally associated with pitch-shifting. Currently, this DSP process is only available in a few graphic waveform editing software programs, such as Passport's Alchemy (developed by Blank Software).

Making plosives ex-plosives. Ah, yes. The dreaded "P"—capable of reducing a beautiful vocal track into a plethora of perturbing pows. But fear not the pernicious plosive. Sampler to the rescue! Here are the details:

1) Once again, sample the entire phrase that contains the plosive. Truncate the sample to ensure there is no data before the onset of the problem consonant.

2) Next, cut a section 1,200 to 1,500 samples long if you're working at 44.1kHz (other sample rates may require cutting a different number), starting 100 samples in from the beginning of the data. To help avoid creating pops and clicks, be sure each end of the cut is at a "zero-crossing." If your sampler doesn't have a CUT function, you can accomplish the same effect by copying the sample, then truncating the original so only the first 100 samples remain, and truncating the first 1,300 to 1,600 samples off the copy.

Create ambience, delay, echo, chorus, and stereo imaging and panning effects without using outboard gear.

3) Next, splice the copy onto the original sample.

4) Finally, roll tape and punch in the new phrase over the original phrase.

Digital punches. Even if your punch-in finger is lightning quick, your tape recorder may not be. How many times have you made what you *know* is the perfect punch, only to find on playback that the machine didn't react quickly enough? Instead of taking chances, try recording the new vocal part directly into the sampler. Next, sample the old vocal part from tape, and splice the new parts into this original vocal. You can edit with accuracy to 1/44,000 second, so you should be able to create undetectable splices.

If you're having trouble making the splice totally seamless, use the CROSSFADE function to blend data at the splice point. This can be especially effective if there are slight tonal differences between the original and the new takes. And don't forget that in the digital world, you can alter a sample's amplitude and frequency content, so don't spend an inordinate amount of time trying to get the vocalist to duplicate a performance. You're never more than a couple of buttons away from a perfect match.

Syncing Up

Throughout this article, we've mentioned recording vocal samples into a MIDI sequencer and synchronizing the sequencer to SMPTE time code. This is the preferred method of working, as it allows you to manipulate the timing of the vocal parts with ease. This is especially critical when you are flying phrases into a lead vocal part, and it can be a real time-saver (and headache deterrent) when you are adding multiple repeating background vocals.

Once you've got your system synced, it's a simple matter to enter the vocal parts into the sequencer at the appropriate time. If you are replacing a phrase, run the tape until the phrase begins, then stop the tape machine. Note the SMPTE time at the location where the tape is stopped. Now simply advance the sequencer to the same SMPTE time and enter in the NOTE-ON event for the vocal part. Roll back the tape several seconds and set the sequencer so it is slaved to the SMPTE code on the tape. Play the tape and listen to the vocal part that is triggered from the sequencer. (The original track should be muted.) If the part plays back too early or too late, simply shift it to a new location. The shift can be done by SMPTE frames, but you'll find that shifting by sequencer clock pulses provides much greater resolution.

The same process is used with repeating background vocals, except it really isn't necessary to note the exact SMPTE location where each part starts. Background vocal events usually can be recorded into the sequencer in real time while the sequencer is synced to the tape.

You may wish to use the sequencer's auto-correct when recording these parts. This minimizes the timing errors that may be induced by less-than-accurate playing. Once you record the parts, you can edit individual note events, or shift the whole track forward or backward in time.

Wrapping Up

If you've never used a sampler for vocals, you might be saying, "Gee, this all sounds like a whole lot of trouble. Besides, I don't have the time to learn all the ins and out of this process." In fact, you don't have the time *not* to. Like it or not, digital technology is changing the way we do things in the studio—and it's changing things for the better. Just look at the advantages: Instant access to any location in a song; unlimited processing and editing; the creation of never-before-heard special effects, and the ability to carry around an entire vocal section on a 3.5" disk!

Once you get the hang of cutting and splicing data instead of audio tape, we guarantee you'll never be able to look at a splicing block and razor blade again. •

The Art Of Manual Noise Reduction

1-Minute Recording Tips

REDUCE ANALOG tape hiss *without using noise reduction.* If you're suffering from tape hiss and don't have (or don't like) noise reduction, this trick may help. First, slightly exaggerate the highs in each signal *before* committing it to multitrack tape. In other words, record each sound source *brighter* than you'll ultimately want it. Then during mixdown, roll off the high frequencies slightly for each track. Essentially, you're applying a form of *manual noise reduction* to your tape.

This also works hand-in-hand with the fact that if you add high-end EQ during mixdown, you end up boosting tape hiss in addition to brightening the recorded signal. So plan ahead and experiment with this "poor person's noise reduction": Boost highs when tracking and cut them during mixdown.

Color-code your patchbay. A Dymo (or similar) labeling gun is great for labeling a patchbay. The problem is having enough horizontal space for both the function and device associated with each jack. Save space by using differently colored plastic tape to signify INPUT and OUTPUT. Color-coding enormously simplifies the process of visualizing signal flow, especially during the heat of a session.

Use double-sided foam tape to mount lavaliers on stringed instruments. A small square of this padded tape is usually enough to attach a lav mike directly to the bridge or soundboard of an acoustic guitar or other stringed instrument. The Countryman Isomax series, by the way, exhibits some PZM (pressure zone micro-

phone) characteristics when placed *capsule down*, within 1/16" or so of a vibrating surface. Many brands of double-sided foam tape are available in this thickness.

PZM-style, or *boundary*, microphones (originally marketed by Crown, now also made by Shure, Audio Technica, and others) lend themselves to all kinds of interesting techniques, since they can be placed on a variety of surfaces (including a floor), and are virtually immune to phase cancellations.

Attention, Alesis HR-16 drum machine owners. Many people use the HR-16 or HR-16B exclusively as a MIDI percussion sound source for external sequencers. Part of the appeal is due to the machine's ability to store 100 drum setups—one per pattern—that include voice settings, tuning, panning, and output level. You also can switch setups using MIDI PROGRAM CHANGE commands from within a sequence.

Some users have been dismayed, however, to find they've lost a carefully prepared setup when they change patches—even after dutifully storing all the parameters as instructed in the manual. It turns out that before the HR-16 can save a setup, it must have at least one beat recorded in the associated pattern. So before you start creating a setup, record a dummy pattern with at least a downbeat into sequence memory. As long as you don't press PLAY or have MIDI CLOCK enabled, you won't hear it.

We're not just whistling "Dixie." If you've ever tried recording someone whistling, you know this ancient art form produces a high volume of air—which is hard to record without the whole thing sounding as if it were recorded in a raging gale at sea. Here's a solution, matey: As the performer whistles, hold a hand out and feel the circle of air pressure being projected. Then position the mike anywhere out-side the pressure area for its best sound. Experiment with several lines of music, since the circle changes with various mouth
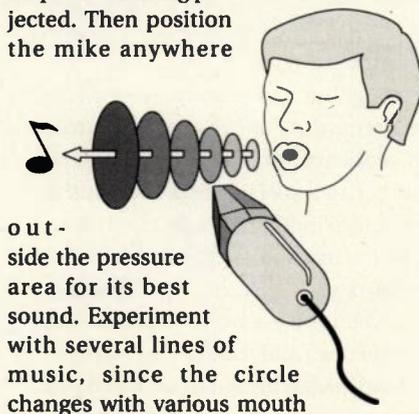
movements. This technique also can be applied to recording pennywhistle, pan pipes, flute, or recorder.

Reduce the risk of tape damage, and extend the life of your tape heads. No matter how new, tape recorder heads all suffer from microscopic-sized pits on the alloy surface that faces the tape. These pits can grab the tape oxide and actually cause damage to your recordings. Furthermore, they increase friction between the tape and the heads, ultimately contributing to head wear. Word has it that TEAC's Stainless Steel Polish can help. This formula, designed to clean front panels and the like, contains tiny silicone particles; a *single* swipe with a swab before each session and cleaning will fill many of the pits on the surface of a head. It's *not* a substitute for cleaning, but it does evaporate quickly, and to our knowledge it has no side effects on heads or tape.

Getting the most from MIDI overdubs. Whether you're Hal Homebrew or Siouxee Superstar, you never have enough simultaneous synth voices and timbres. You want the best of both worlds—big sounds from layering synths and as many sequenced tracks as possible.

The answer is to use sync-to-tape in a versatile way. Before rolling tape, build up your composition in the sequencer using only one sound per track, recording as many tracks as possible (or necessary). Stripe the tape with your sync code before you actually record anything to multitrack. Finally, solo each sequencer track and record it in sync to a unique tape track—"channelizing" as many MIDI synth layers as necessary to get a humongous sound for each one. Repeat the layering process with unique sounds for each pass.

We'll pay for your tips! Have you ever found a $4 gizmo at a hardware store that saved you hundreds of dollars in studio time? Ever stumble on a great recording technique that would make George Martin envious? Or discovered what an owner's manual didn't tell you after staying up all night? How about the time you built that MIDI-controlled toaster?

If we publish your studio solution, we'll tip you $25 to $100 (depending on the complexity of the subject). Photos and sketches are worth bonus points. (Sorry, but we can't be responsible for returning submitted materials.) Send your tips to: Studio Solutions, EQ magazine, 20085 Stevens Creek Blvd., Cupertino, CA 95014.

Harmony Without Tears

Circle Of Fifths

IF YOU RECORD MUSIC, it's important to have a grasp of basic harmony. Using a device known as the *cycle chart*, or Circle of Fifths and Fourths, you can get a handle on this subject in no time at all. Not only will you be able to transpose a chord progression at will, you'll communicate more effectively with studio players. And, if you're a songwriter in search of the proverbial "lost chord," or a producer co-writing with your artist, you'll appreciate how the cycle chart quickly reveals the essential harmony of every key. As a bonus, you'll immediately see which chords from outside the key will lend harmonic spice to a progression. Here's how it works:

The Cycle Chart. Letters representing the 12 major keys of Western music are arranged around a circle (see adjacent figure). When moving clockwise, each letter is separated by a perfect fifth; traveling counter-clockwise, each move corresponds to a perfect fourth. Key signatures (a key's particular complement of sharps or flats) are shown immediately inside the circle.

Using the cycle chart, you can determine the primary *triads* of any key. First, target a fundamental *keynote*; this represents the I chord. From the I chord, move one position clockwise to locate the V chord (a fifth away), and one position counter-clockwise for the IV chord (a fourth away). In the key of *Bb*, for example, *Bb*, *Eb*, and *F* are the I, IV, and V chords.

Every major triad has a *relative minor*. These triads appear inside the circle, across from the key signature that they share with their relative major. Like the IV and V chords, relative minors are numbered according to their distance from I. (Upper case Roman numerals indicate major chords; lower case, minor chords.) The relative minor of I is vi (a major sixth away); for IV it's ii (a major second away from I) and for V it's iii (a major third away from I). In the key of *Bb*, the ii, iii, and vi chords are *Cm*, *Dm*, and *Gm*. Looking for effective chord substitutions? Try replacing a major chord with its relative minor.
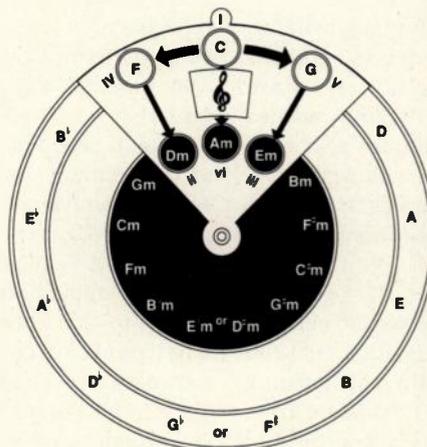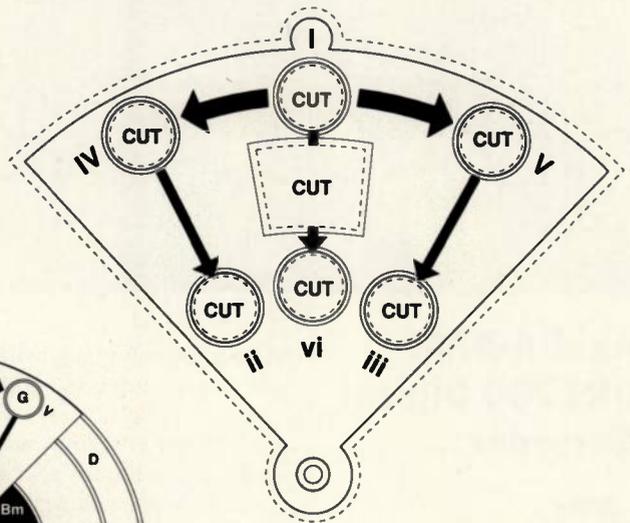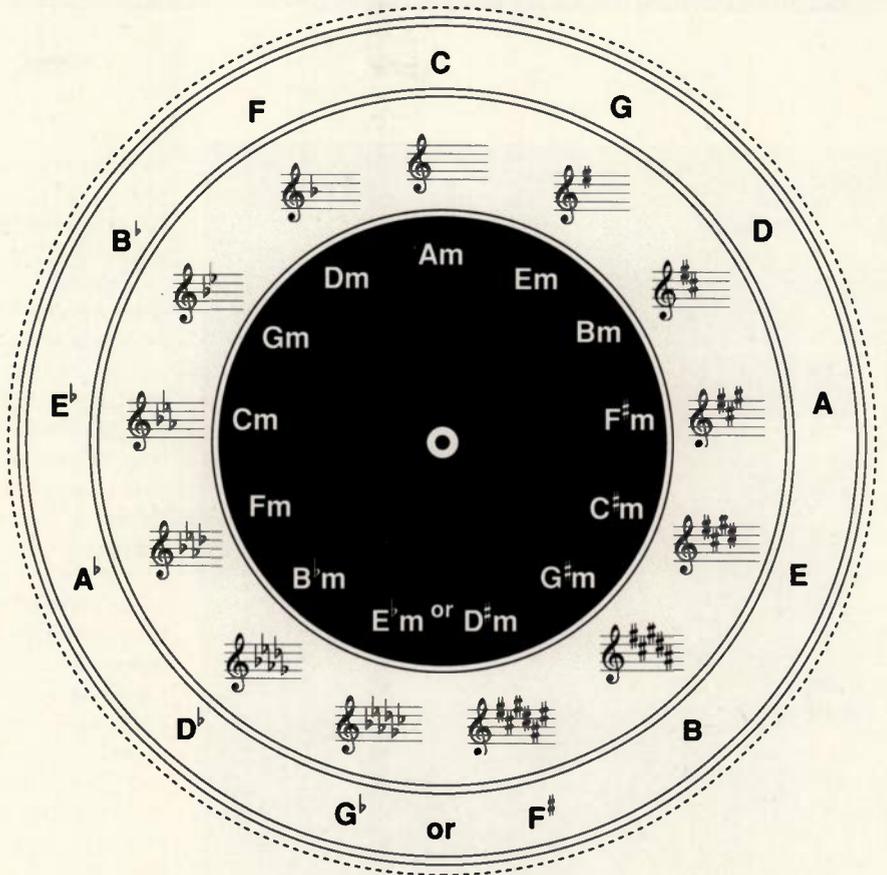
Here's how to create an instant (and almost free) transposition tool: Photocopy this page, and glue it to a piece of light cardboard. Cut out the big wheel and the chord function template. From the template, carefully remove the insides of the seven viewing holes (marked "cut") with an X-Acto knife. Finally, using a brad, affix the template to the wheel, as shown in the small assembled diagram. Simply rotate the template to determine the key signature and primary chords for any of the 12 Western music keys!

Akai A-DAM DR1200 Digital Recorder

DIGITAL multitrack tape recorders are wonderful. At approximately $4,000 to over $6,000 per track, however, today's 24-, 32-, and 48-track digital recorders are not cheap.

Enter Akai with their A-DAM DR1200, an expandable, digital multitrack recording system. Of course, the term "affordable" means different things to different people, but at $2,000 per track (for the 12-track base model), the DR1200 smashes some price barriers. It also smashes some technical barriers: It's the world's first rotary-head digital *multitrack*.

OVERVIEW

The A-DAM system consists of three units: the DR1200 Digital Audio Multitrack Recorder, DM1200 Meter Unit, and DL1200 Programmable Autolocator. A-DAM is an acronym for Akai Digital Audio Multitrack Format. (Who knows where the "F" went? Our guess is that A-DAMF just doesn't read too well.)

The unit is compact, but not exactly portable; it weighs in at a hefty 80 pounds. The main unit houses the tape transport and all the digital and analog interfaces for the recording system. Data is stored on 8mm video cassettes; a thoughtful choice on Akai's part considering the wide availability and low cost of these tapes. (Akai may have learned its lesson when the proprietary-format cassette tapes used for its 12-track analog system received a less-than-enthusiastic response from some.)

The compactness of the main unit can be attributed in part to the use of eight custom large-scale integration chips (LSIs), which contain the entire PCM system. Photocouplers and multiplexed logic are used to help prevent problems such as electromagnetic and electrostatic leakage between digital and analog systems, thus avoiding induced noise.

Recording Format. DR1200 provides two sampling frequencies, 44.1kHz and 48kHz, with 16-bit linear quantization. The tape speed is 2.862 ips at the 44.1kHz sampling rate, and 3.094 ips at the 48kHz sampling rate. Using a P6-90 (NTSC 90-minute) tape, up to 17 minutes of data can be recorded at 44.1kHz sample rate; 15.5 minutes at 48kHz. Sampling error correction is performed by using a Double Reed-Solomon code with channel scrambling. This type of error correction can recover/correct an error of up to 3mm in length on the tape. A recording called MAINTENANCE may be made to check for errors. When this tape is played, a front-panel indicator illuminates if too many errors are encountered.

In addition to the 12 digital tracks, there are two analog tracks—one recordable for time code and the other for an internal control track.

THE SYSTEM: A CLOSE LOOK

The Transport. The tape transport uses a new U-loading mechanism that provides smooth transport characteristics and reduces wear and tear on the tapes. The front-loading tape mechanism is much the same as the one used in video and DAT machines. The unit sports a rigid die-cast aluminum chassis, which results in a reliable tape transport. (Previously, aluminum chassis were found only on the larger, more expensive stationary-head digital recorders.)

One advantage of a rotary-head digital recording system is that wow, flutter, and speed drift—common problems with analog tape transports—are inaudible. The A-DAM's speed stability is enhanced by matching the internal clock signal against a precision crystal oscillator on playback. Additionally, the tape speed is servo-adjusted to eliminate errors.

The rotary heads and the headblock assembly are accessible from the top of the main unit, allowing head cleaning and/or replacement to be performed easily. Three Akai rotary digital record heads and three Akai digital playback heads are held on a single, helical-scanning head drum that rotates at 2100 rpm. There are also two fixed analog record/playback heads (for time code), one analog erase head, and one full erase head. Akai reports that the average life-span of the heads is

about 1500 hours. This is comparable to the length of time that analog tape heads can be operated before they require service or replacement. To reduce head were, the A-DAM will retract the tape from the heads automatically after two to three minutes of inactivity.

Unfortunately, the Akai heads and drums are not replaceable in the field. The entire headblock must be removed, and that procedure requires special tools and training. Head cleaning, which must be performed to minimize digital errors, is also a sensitive procedure, as we'll learn a little later in this story.

The front of the main unit has several indicators including CASSETTE IN, TAPE RUN (which shows that the tape is moving), EMPHASIS ON, SAMPLING FREQUENCY, TAPE VIEW (which allows you to see the cassette), INTERNAL OR EXTERNAL SYNC 1 OR 2, UNFORMAT, and REC PROTECT. These indicators provide a quick means of checking the A-DAM's status, and can be helpful when troubleshooting the system.

Making The Connection. The main unit's rear panel contains connectors for interfacing the DR1200 to external equipment. The DIGITAL IN/OUT jack can be used to connect the unit to other DR1200s or to the optional AES/EBU DF1200 digital I/O module (scheduled for release in early 1990), which will allow the DR1200 to interface digitally with equipment such as DAT recorders. The DIGITAL LEVEL METER jack connects the DR1200 to the DM1200 meter bridge. The SYNCHRONIZER jack connects the unit to an external synchronizer, for slaving the DR1200 to a VCR or another audio recorder. The LOCATOR port is for the DL1200 autolocator.

Up to three DR1200s can link together via the SYNC IN/SYNC OUT connectors, allowing you to configure a 24- or 32-track system. All the connectors mentioned above are Amp D-Sub connectors ranging in size from 25 to 37 pins. Two 1/4" unbalanced phone jacks marked AUX IN/AUX OUT are used for analog audio tracks.

The DR1200's rear panel also provides connectors for the analog audio outputs of the digital audio tracks. They're low impedance, balanced, 3-pin XLRs (male/female) for interfacing with audio mixers. The XLRs are wired with pin 3/hot, pin 2/common, and pin 1/ground (useful to know when deciding on what cabling to use with the DR1200, since there still is no industry-wide agreement on the adoption of pin 3/hot or pin 2/hot as a standard).

Meter Mania. The system's second component, the DM1200, is a meter unit designed for use with the DR1200. We really appreciated the sophistication of A-DAM's metering system. First-time users should find it enables them to get the most out the system without the level problems usually associated with digital recording.

The meters include a 16-bit 0dBm reference level, a 15-segment digital level meter for each of the 12 digital audio tracks, and a 12-segment level meter for the AUX (analog) channel. A single cable connects the meter bridge to the main unit, and power is provided by the DR1200.

Each vertical bar meter in the DM1200 performs three functions: continuous peak level display; peak/hold level display; and channel record status. At the bottom of each column of

LEDs, there's a red bar that illuminates when the correspondingly numbered channel is in RECORD mode. The rest of the LEDs in the column are labeled to indicate the input or output level (depending on the selected mode) for the channel in dB, calibrated to 0dB = 0dBm.

Peak levels are particularly significant in digital recording. In analog recording, a signal that exceeds the allowable level (the clipping point of the electronics or the saturation point of the tape) creates some degree of harmonic distortion. If the tape is saturated, the distortion won't be terribly objectionable to the ear. But in a digital recorder, the *absolute* level recorded on tape is always the same. The level

that comes out of the XLR audio outputs, however, varies according to the digital information retrieved from the tape. Thus, you might think a digital system is less susceptible to excess level, but that's not the case. The analog-to-digital converter can convert levels only within a specified range of levels and frequencies.

We found the A-DAM sensitive to excess audio levels, particularly at very high frequencies. Excess high-frequency energy results in a unique type of distortion known as aliasing, which is unnatural-sounding and unpleasant to the ear. For this reason, it is important to observe the peak levels and make sure they do not exceed

EQ Background Notes

Rotary-Head PCM Digital Recording

For close to ten years now, PCM digital encoder/decoders have been available from Sony, Technics, and others. Using a video cassette recorder as the storage device, these devices (such as the Sony PCM-F1) record and playback digital audio as if it were a video signal.

The PCM encoding process samples incoming analog sounds, typically as often as 48,000 times per second. A digital numeric code is assigned to each sample; as sounds change, so do the samples. The codes get stored during recording, and get decoded during playback. PCM stands for *pulse code modulation*, since each sample is like a *pulse* being registered, and the codes change with the pulses.

VCRs have *rotary heads*, as do DAT recorders and the Akai A-DAM. Rotary heads provide certain advantages over the traditional stationary heads found on almost all analog recorders. A drum with the record/playback heads spins at high speed against the slow-moving tape. The results are: a faster apparent head-to-tape speed that yields a higher bandwidth; a very stable transport with virtually no wow-and-flutter; and less tape must be used.

But rotary heads aren't perfect. They don't allow for razor-blade editing. Instead, sophisticated electronic editing is required for digital audio, at far greater resolutions than the sub-frame accuracy required by video counterparts. This is expensive. And stationary-head digital technology still requires designers to jump through fewer flaming hoops to perform editing, punching, and simultaneous multitrack record and playback in sync.

—*Jeff Burger*

the maximum values, which are indicated by red lines.

While momentary peaks are capable of causing audibly detectable distortion, such peaks may not be noticed on the meter due to their brief, transient nature. So the DM1200 is equipped with a peak hold circuit. When the PEAK HOLD button is set to the AUTO mode, each channel's highest peak level will be displayed for about 1.5 seconds—enough time for the operator to see what the peaks are doing. The AUTO mode is well-suited for general-purpose use. We found the CONT mode (PEAK HOLD button disengaged) more useful when we were establishing mix levels or involved in a particularly busy mix session where there were lots of distractions. In the CONT mode, the highest peak is continuously displayed. This lets you see peaks that have exceeded the allowable value, even if you were not looking at the meter when they occurred.

Two variations of the AUTO mode are provided: PEAK I and PEAK II. These let you select the release time—319ms or 1277ms—that is used for the peak display. The length of time a peak is displayed is determined by three factors: the peak's duration; hold time; and release time. The hold time—about 1.5 seconds—occurs after the circuit no longer senses the peak's presence. The release time begins immediately at the end of the hold time. Therefore, the total time the peak indicator is illuminated is equal to the peak's duration, plus about 1.5 seconds, plus the release time.

The Autolocator. The DL1200 programmable autolocator unit is the third component of the A-DAM system. The DL1200 is used to control the tape transport mechanism and

EQ Lab Test

Digital Audio Channels
Frequency Response:
Claimed: 20Hz—20kHz (+1.0dB, -1.2dB) @ 44.1kHz sampling rate.
Tested: 20Hz—20kHz (+0.9dB, -0.8dB) @ 44.1kHz sampling rate. 20Hz—20kHz (+0.6dB, -0.5dB) @ 48kHz sampling rate.
Signal-to-Noise Ratio:
Claimed: 90dB or greater with emphasis on.
Tested: 87dB or greater with emphasis on @ 44.1kHz, NAB weighted.
92dB or greater with emphasis off @ 48kHz, NAB weighted.
Distortion:
Claimed: 0.05% or less THD (Standard level = full scale less 15dB).

Tested: 0.06% or less THD (Standard level = full scale less 15dB).
Crosstalk:
Claimed: 80dB or better @ 1kHz @ 44.1kHz sampling rate.
Tested: 85dB or better @ 1kHz @ 44.1kHz sampling rate.
90dB or better @ 1kHz @ 48kHz sampling rate.
Wow & Flutter:
Claimed: Beyond measurable limit.
Tested: Beyond measurable limit.

Analog Audio Channel
Frequency Response:
Claimed: 100Hz—14KHz (±3dB).
Tested: 75Hz—15kHz (±2.5dB).

to switch channels between READY, RECORD, and PLAYBACK modes. The autolocator also has a number of versatile memory functions including auto punch-in/out and pre/post roll, which can be used for rehearsals prior to actual recording of a take. The location and tape counter display resembles a SMPTE time code read-out with HOURS, MINUTES, SECONDS, and FRAMES. (Akai tells us this is a reading of the machine's control track, and is not a true SMPTE frame-accurate code.)

The DL1200 is designed for ease of use, with many functions requiring only a single button to be pressed—as opposed to the two or three button pushes required on other autolocators. In total, the DL1200 provides 69 keys for accessing settings and functions. If you're thinking of going for a full-blown 36-track A-DAM system, you'll be happy to know that a single DL1200 can control three DR1200 recorders simultaneously.

The operator can fully automate the complicated operations necessary in multitrack

recording, using the LOCATE, DIRECT, REPEAT, RETURN, PREROLL/POST-ROLL, and other DL1200 functions. AUTO PUNCH IN/OUT provides a quick, easy method for recording multiple takes over the same part of the tape. The operator can check the punch points without actually recording on tape, using the REHEARSAL function. When this mode is used, the monitoring switches from playback to input, as though actual recording were taking place.

Nice Touches. The DL1200 also offers a number of convenience features accessible from the autolocator, including a CHANNEL PRESET function that lets the operator immediately recall one of four recording channel mode settings. Five direct-address time memories and an additional 100 numerically addressable stack memories allow you to define 105 different cue points on a single tape. You can use the memory locations for such functions as locating, repeating, and autopunching.

Recorded material can be reproduced (played back) with a programmable delay (from

zero to 66 milliseconds). The amount of delay can be set independently for each channel. A CROSSFADE function lets the operator set a crossfade time for REC IN and REC OUT of up to 92.8ms, so the transition of sound produces no audible glitch. (In the analog world, this utility would be similar to gapless/seamless punch-in and -out.) Pitch may be varied in 0.1% increments between –6% and +6% of the nominal tape speed using the PITCH CONTROL function, but this can be performed only on material that has been recorded using the 44.1kHz sampling rate.

One interesting feature of the A-DAM format is that a 30-second TABLE OF CONTENTS (TOC) area is included at the beginning of a tape. This allows the channel presets, time memory, stack memory, delay, and other memory data to be saved with each tape. Every time you go back to a previously recorded tape, you can reload all the appropriate data simply by loading the TOC data. This means you can retain all your settings when power to the system is shut down—provided you've saved your data to tape.

USING THE A-DAM

Assembling the entire system, and getting it up and running, turned out to be breeze. The ease of operation is due—in no small part—to the comprehensive manual included with the unit. It's one of the best "here's-how-to-use-it" manuals we've seen to date.

The supplied cables are long enough for the main unit to be placed a good distance from the operator, while keeping the metering unit and the autolocator within operating distance. A single power cable provides power for all three units. All the connector cables

are keyed, so it's impossible to insert a cable into the wrong port.

The DR1200's fan and motor noise is relatively quiet, generating an average room noise level of -45dBA. However, we did detect some squeaking from the rotary head that sounded as though some SMPTE code was being generated. We eliminated the noise by making a simple adjustment to the metal guide on the top of the rotary head.

25¢ To Happiness. We undertook the most delicate part of the DR1200 system's operation: cleaning the heads and tape guides. Akai warns you never to use 8mm head-cleaning cassettes as they are abrasive, can shorten the head life, and can cause damage to the drum.

We gained access to the rotary and fixed heads, as well as to the capstans, pinch rollers, and guides using the Special Opening Tool—a genuine United States 25-cent piece. The rotary head is delicate and should be cleaned with absolute ethyl alcohol (which contains no water or oil) and a Kim Wipe lint-free cloth. Akai informed us that Freon TF was also acceptable for use on the heads. [*Ed. Note: Due to environmental considerations, we do not recommend the use of Freon TF.*]

The DR1200 manual says that cotton swabs are fine for cleaning the fixed heads and the pinch rollers and guides, but as with any digital system, sponge-type cleaning swabs (like the ones commonly used on VCRs) are preferred.

Power On. When power is first applied to the main unit, a diagnostics program runs for about 30 seconds, after which all of the indicators illuminate. If a problem occurs, the warning indicators on the main unit

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Product: Akai A-DAM digital multitrack tape recorder system
Manufacturer: Akai. Distributed in the U.S. by IMC, 1316 E. Lancaster St., Fort Worth, TX 76113; (817) 336-5114.
Suggested Retail Price: 12-track system, $24,999.00. 24-track system, $45,500.00. 36-track system, $66,000.00.
Tape Format: 8mm video cassette
Record Time: P6-90 (NTSC 90-minute) tape, up to 17 minutes at 44.1kHz sample rate, 15.5 minutes at 48kHz.
Recording Format: 12-track PCM digital + 1-track analog + 1 internal control track. 44.1kHz and 48kHz sampling rates. 16-bit linear quantization. Error correction: double Reed-Solomon code with channel scrambling.
Interfacing: Amp D-Sub connectors for digital in/out, digital level meter, synchronizer, locator, sync in, sync out. Two 1/4" unbalanced phone jacks for aux in and aux out. 12 male, 12 female low-impedance, balanced, 3-pin XLR audio jacks.

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and on the autolocator illuminate.

One time, the WARNING light came on when we powered up the unit after it had been sitting on our bench for a week. We determined that this was due to the unit being very cold, and to condensation on the heads. The condensation was due to cold weather and our proximity to San Francisco Bay. (It's foggy in these parts, you know.) Akai recommends you let the unit warm up for 30 minutes before using it, or 60 minutes if the operating environment is very cold. The manufacturer's specifications state the operating range as 10°C to 35°C (50°F to 95°F). Apparently Akai knows their specifications: When we tested

the units in a cold environment and let them warm up for only a few minutes, our test results were worse than those obtained under the recommended operating conditions.

Pick A Rate. A-DAM system tapes are similar to computer hard disks in that they must be formatted before anything can be recorded on them. We used TDK Video 8 60-minute cassettes. To save time and trouble, we recommend you format several cassettes before you start a project. Prior to formatting the tapes, you must choose the 44.1kHz or the 48kHz sampling rate, and decide if you are going to use emphasis—which boosts some of the higher frequencies.

Once you've made a choice, there's no turning back. Later, if you decide to change the sampling frequency or emphasis mode, you'll have to reformat, and the formatting process erases all previously recorded data. We formatted four tapes for our tests, two each at 44.1kHz and 48kHz, one set with emphasis, the other without.

How Does It Sound? We put a 12-track A-DAM system to the test during a recent recording session. Several different listening tests were performed. For the first test, we recorded a CD into the A-DAM and compared the CD playback with the A-DAM's playback of the same material. For another test, we transferred R-DAT masters to the Akai system and compared the original source with the A-DAM's output.

The general consensus of our panel of experts was that there is a difference in sound quality between source materials and A-DAM's reproduction of those materials. On the whole, the A-DAM seemed a little raspier in the high end and a little tubbier in the low

end. (We apologize for the less-than-scientific adjectives). Since the A-DAM is not a 3-head analog recorder, and consequently doesn't let you "A/B" the source input and the recorded material, it was difficult to determine if the differences we heard were a result of the methods we used to input the source material, or the result of coloration induced by the A-DAM system itself.

To create a more accurate comparison test, we striped the AUX (analog) channel with SMPTE time code, then used A-DAM as a code-only SMPTE master to drive an NED 16-track, Direct-To-Disk Synclavier system. By punching in and out on the A-DAM, we were able to compare a song recorded on the Synclavier with the signal reproduced by the A-DAM. Once again, we perceived a minor difference between the original input source and the output of the A-DAM.

(A side note: The A-DAM's AUX track is always in repro mode, so whenever you rewind, the slave machines will follow—providing you have a high-speed time code reader. Akai furnished us with a diagram of how to build a synchronizer control cable for the Adam/Smith Zeta-Three SMPTE module so the A-DAM could be used as a slave unit, but we weren't able to construct the cable in time for our editorial deadline.)

We also used the recording session as an opportunity to put the autolocator through its paces. All the functions, including PRE-ROLL, POST-ROLL, AUTO PUNCH-IN/OUT, and REHEARSAL, worked like a charm. After just using the Synclavier system, which locates immediately to any specific point, the A-DAM sometimes felt a little slow when returning to far-

h.dd e n a s s e t s

Yamaha DD-5 Rhythm Box

Undocumented Features Reveal A Pad Controller Disguised as Toy

The Yamaha DD-5 may be the world's cheapest MIDI controller. Ostensibly it is a "music hobbyist's," non-programmable rhythm box that sells for under $100 at consumer electronics, toy, and department stores. We bought ours on sale for $67.

The DD-5's list of features includes 30 preset rhythms, 12 PCM sampled sounds, four pads, and a built-in speaker. That's just on the surface. Upon further inspection, we discovered a functional MIDI OUT, and found that the pads are velocity-sensitive. Amazingly, these pads transmit MIDI NOTE ON messages at 15 velocity levels (see Fig. 1). Although this is undocumented in the cryptic manual that accompanies the DD-5, you can program a pad to send any MIDI note # of your choice. Not bad for a device you might buy for a child's birthday! Here's a step-by-step description

of how to maximize the hidden potential of the DD-5. The DD-5 isn't a $900 Roland Octapad, but with a bit of patience, you'll find it to be a remarkably powerful tool.

Hidden Defaults. To cycle through the 12 samples, hold the PAD ASSIGN button and tap a pad. Each sample has an associated MIDI NOTE # (Fig. 2). With every hit, this MIDI NOTE # is transmitted as part of a note on message via the MIDI OUT port. When you power up, the pads default to the MIDI NOTE #s shown in Fig. 3. As you step through drum samples on a pad, the MIDI NOTE #s change accordingly. Assign a sound, and you automatically assign a MIDI NOTE #. For example, the top left pad

defaults to MIDI NOTE # 57 (sample 9, hi hat closed). To assign it MIDI NOTE # 52, cycle through five samples (10-2), arriving at the snare. At the DD-5's minimum volume setting (there are only three levels), its samples remain audible. So when you're finished assigning MIDI NOTE #s and are ready to trigger an external device, mute the internal speaker by inserting a 1/4" phone plug into AUX OUT. If you're using AUX OUT, drop the fader at your mixer.

Secret Procedures. Need more than 12 MIDI NOTE numbers for sequencing? One approach is to remap the defaults at your sequencer. Alternatively, you can reprogram the pads to MIDI note #s 0-127. (Warning—these settings are lost when you power down.) Here's how it's done.

Tap any pad while holding PAD ASSIGN and pressing VOLUME UP; all four pads shift to MIDI NOTE #10. (volume down resets all pads to

MIDI NOTE #120.) From either of these global settings, change each pad individually as follows: Hold PAD ASSIGN, press VOLUME UP or DOWN, and tap (each press increments or decrements by 10 MIDI NOTE #s respectively). Advance MIDI NOTE # units (+1) by holding PAD ASSIGN and tapping.

If you get lost, re-initialize all pads by holding PAD ASSIGN and pressing VOLUME UP and DOWN simultaneously. Rather than work "blind," monitor your progress with a sequencer that displays incoming MIDI NOTE #s. In a pinch, use a synth keyboard to provide audio feedback of MIDI NOTE # assignments.

• • •

So the DD-5 doesn't offer programmability, doesn't transmit all 127 velocity values, has only four pads, and lacks a bracket for stand-mounting. But if you're looking for a supplemental controller to sit alongside your professionally equipped Octapad—or simply want something that suits a tight budget—the DD-5 is truly a bargain. You may even like some of its mediocre built-in sounds. And if you're still not convinced, consider this: Yamaha even throws in a cheap pair of drum sticks!

—Andy Widders-Ellis

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away tape positions. Still, the transport is as fast as those on typical analog recorders, and we found the A-DAM was quite precise when locating cue points.

One test we performed was an old analog test of gapless/seamless punch-in/out capability. A 1kHz tone was recorded, the tape was rewound, then a new signal was punched in and out at various times on the tape while the

The A-DAM system is a good production tool for a compositional workshop, and would be right at home in a pre- or post-production video/film studio.

original tone was played back. We were able to adjust the CROSSFADE times on the autolocator so the punch-in/out areas were barely discernable. When we tried the same test on the Synclavier, we obtained virtually identical results. In both systems, we consider the results to be excellent.

One final note: We found that the best-sounding recordings were done at the 48kHz sampling frequency with no emphasis. Next-best was

44.1kHz with no emphasis. Again, we were performing subjective listening tests, but to our ears, the recordings sounded harsh and brittle when emphasis was used.

CONCLUSIONS

We came away with a short wish list for the makers of the A-DAM:

• Find a tape formulation that yields more than 17 minutes of record time per cassette.

• Supply pre-made SMPTE control cables for external synchronizers.

• Allow real-time SMPTE

numbers to be displayed in the autolocator windows.

• Allow variable pitch control at the 48kHz sample rate.

• Improve the overall sound quality when emphasis mode is used.

• Design a field-replaceable headstack.

All Things Considered. The Akai A-DAM digital multitrack is compact, features excellent tape-handling characteristics, and offers very good sonic reproduction. As if these things weren't enough, the arm rest on the autolocator is one of the finest ever seen. (Hey, now we're talking about important

DD-5 MIDI VELOCITY RESPONSE TABLES

VELOCITY 1	LEVEL	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
	MIDI	15	23	31	39	47	55	63	71	79	87	95	103	111	119	127
VELOCITY 2	LEVEL	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15
	MIDI	16	24	28	32	36	40	44	48	52	56	60	64	72	80	88

	SAMPLES	MIDI NOTE #	NOTE/OCTAVE
1	BASS DRUM	45	A1
2	SNARE DRUM	52	E2
3	TOM 1	53	F2
4	TOM 2	50	D2
5	TOM 3	48	C2
6	RIDE	63	D#3
7	CONGA	65	F3
8	HH OPEN	59	B2
9	HH CLOSED	57	A2
10	RIM	51	D#2
11	COWBELL	55	G2
12	HAND CLAP	54	F#2

Fig. 1. (above) The DD-5's four pads transmit out 15 different MIDI velocity levels. Fig. 2. (left) MIDI NOTE NUMBERS for the 12 sounds. Fig. 3. (below) MIDI NOTE NUMBER defaults for the pads.

PAD	MIDI NOTE #	NOTE/OCTAVE
UPPER LEFT	57	A2
UPPER RIGHT	63	D#3
LOWER LEFT	52	E2
LOWER RIGHT	45	A1

Wanted: Hidden Assets

We're looking for diamonds in the rough. If you know of some device with useful features that aren't documented in the manual or are generally overlooked by most users, let us know. We're especially pleased to learn about devices that perform much better—or accomplish much more—than their prices would suggest, although price is not our primary consideration. The device can be well-known or unknown. Affordable modifications or enhancements to an existing device are also of great interest to us.

We'll pay from $35.00 to $100.00 for any detective work we choose to publish, depending upon length, content, illustrations, and usefulness of the information. Send it to Hidden Assets, EQ Magazine, 20085 Stevens Creek, Cupertino, CA 95014. Tell us the name and manufacturer of the product, and document its hidden assets. Also tell us what steps, if any, must be performed, and include diagrams if they help explain the process. Sorry, we can't assume responsibility for the return of articles or artwork not assigned by our staff.

stuff!) As the capper, the system's price makes it hard to resist. Even if you decide to expand to the full 36-track system, the investment/capability ratio is hard to beat.

We do have some concerns about the long-term effects that rotary heads may have on tapes. Although we've been using our DAT deck for 2-1/2 years with no noticeable tape degradation, DAT tapes are not shuttled back and forth during record and playback as many times as tapes would be on the A-DAM system. Unfortunately, any potential problems in this area will be revealed only after

a fairly long period of operation. Our feeling is that if A-DAM offers the capabilities you need, you shouldn't worry too much about tape wear.

The A-DAM system is a good production tool for a compositional workshop, and would be right at home in a pre- or post-production video/film studio. If you need a high-quality—yet affordable—multitrack digital recorder, the A-DAM is here, and it delivers.

—*Richie Moore, Ph.D.*

DigiTech DSP256 Multi-Effects Processor

THE MARKET for multi-effects processors continues to expand as features increase and prices plummet. One of the latest entries, the DSP256 from DigiTech, sports dozens of notable improvements over their original DSP128, including twice the memory, ten additional effects configurations, and a 15-button, full-function, wired remote controller designed for easy use in the studio. The DSP256 also boasts a new, cleaner look, complete with a back-lit LCD display (2 lines x 16 characters) that displays programs and edit parameters with full or slightly abbreviated names. There's also a 3-digit LED PROGRAM NUMBER display that's large enough to be seen across a control room.

Factory programs. The DSP256 is shipped with 128 factory presets in ROM and 128 user-programmable memory locations in RAM. The factory presets make use of nine basic functions: STEREO CHORUS, STEREO FLANGE, STEREO DELAY, 4-TAP DELAY, ULTIMATE REVERB, GATED REVERB, REVERSE REVERB, a 3-band PARAMETRIC EQ, and a 9-band GRAPHIC EQ. These functions are available alone or grouped into 16 additional "configurations" of up to three simultaneous effects plus an associated programmable mixer function. Since the mixer's ability to configure effects in a parallel or series manner—or as a mixture of the two—has a direct bearing on how the effects sound, you could think of the mixer as a fourth simultaneous effect.

All individual effects parameters are available when using combined effects—very nice. A welcome improvement over the DSP128 is the ability to scroll forward and backward through the parameter list when editing programs and changing values.

The factory presets sound great. They are well thought-out, clean, and transparent, and most are immediately usable. Many employ the ULTIMATE REVERB configuration. In addition to the early reflection parameters that are fairly standard on reverb devices, this configuration provides subsequent parameters for precise control of the reverberation characteristics after the first or initial reflection occurs. One factory program that uses these features, MARBLE MANSION, is absolutely gorgeous.

MIDI. The DSP256's MIDI functions are comprehensive. Each of the 90 programmable parameters can be controlled by any one of 128 MIDI controllers or MIDI CHANNEL PRESSURE. The PATCH MAP allows any external MIDI PROGRAM CHANGE command to recall any desired internal program. The DSP256 also can transmit MIDI PROGRAM CHANGE messages to other MIDI devices. On top of that, the entire contents of memory can be transmitted and received in a MIDI data dump (useful for storage and editing with a computer, or for reprogramming another DSP256).

Details. The operation manual is fairly well-written. The inclusion of block diagrams of each configuration really helps the user understand the signal flow. Also noteworthy is the unit's silent switching in or out of BYPASS mode. The detachable AC cord is good news if you despise those external power supplies which usurp additional space on AC

strips. The construction is extremely solid. Internally, we were impressed to see socketed ICs, fiberglass PC boards, and clean wiring.

Inputs and outputs. The DSP256 operates with a wide variety of instruments and mixers. It can function as a preamp for guitars and basses (which operate roughly at a -20dBu level), or it can receive -10dBV ("semi-pro") or +4dBm ("pro") levels from a mixer. The audio jacks are all 3-conductor balanced 1/4" phone; the unit works fine with balanced or 2-conductor unbalanced plugs. As with most multi-effects units in this price range, the stereo inputs are summed to mono during processing; the effects algorithms simulate stereo at the left/right outputs.

Under Scrutiny. While the DSP256 has no trouble operating at +4dBm levels, our tests did find the unit to fall short of the specified maximum input and output levels (see sidebar). In real-life terms, this shortcoming will probably *never* affect most users—although it does mean that the input level meter should be watched carefully for overload, particularly when used with a +4dBm source such as a recording or P.A. mixer. (For broadcast applications in a studio wired at +8dBm levels, one is advised to use a level-matching interface with the DSP256.)

INPUT and OUTPUT LEVEL settings are variable via front panel knobs, as is the MIX control. These are handy, and are sorely missing from some other multi-effects devices in

this price range. MIX LEVEL is also a programmable parameter for each effects patch.

Our biggest disappointment came when we learned that the BYPASS function is not a true bypass. In fact, with the MIX knob in any position other than completely DRY, signals pass through the digital stages of the DSP256 even with BYPASS engaged. They don't receive any reverb or other effect, but they do get delayed slightly by the process of converting the incoming analog signal to digital and back to analog.

Normally this wouldn't be a problem. In fact, if you're using the unit as a preamp for a guitar or other instrument, it's a good thing that the BYPASS doesn't short the inputs to the outputs—otherwise you'd suffer a drop in your signal. But this design approach has drawbacks. When the manual MIX control is set at a 50:50 ratio between WET and DRY (and with BYPASS engaged), some serious phase cancellation occurs, since the unit is combining dry signals with delayed, but unprocessed, wet signals. As our test results indicate, this resulted in wild frequency variations. In the NON-BYPASS (active effects) mode, these variations were less severe, but still problematic. With MIX in the full WET position, in BYPASS or NON-BYPASS

modes, the frequency response was much better, though nowhere near the specs claimed.

If you're using the DSP256's BYPASS button to compare processed and original signals, it's less likely you'd hear these frequency deviations in a live setting. In the studio, however, they could be problematic. Fortunately, there's an easy solution: Don't use the unit's BYPASS button to compare. Instead, simply mute the returns from the DSP256 at the console (or bring down their level). This may be an inconvenience for some users, but it's a reasonable one.

Since we encountered troublesome (although lesser) phase cancellations with a 50:50 mix in the NON-BYPASS mode, we recommend that the DSP256 be operated, if possible, with the MIX control set to the WET. This is proper engineering practice in almost all

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EQ Lab Test

Product: DSP256
Manufacturer: DigiTech, 5639 S. Riley Lane, Salt Lake City, UT 84107; (801) 268-8400
Price: $529.95 including remote.
Frequency response (BYPASS mode):
Claimed (wet): 20Hz—20kHz (+0dB, -3dB).
Tested (dry): 20Hz—40kHz (±0.4dB).
Tested (50:50 wet:dry ratio): ±1dB to 1.45kHz; down 24dB at 3.1kHz; flat at 6kHz; down 19dB at 9.4kHz; flat at 13kHz; down 13dB at 15.5kHz; flat at 17.5kHz; gentle roll-off to 6dB down at 37kHz. These severe variations are induced by phase cancellation—see story.
Tested (wet): Down 2dB at 20Hz; ±0.5dB from 185Hz to 2.7kHz; rolling off 2dB per octave to a maximum of -6dB at 17.5kHz; signal drops off dramatically after 17.5kHz.
Frequency response (NON-BYPASS mode):
Claimed (wet): Same as above.
Tested (dry): Same as above.
Tested (50:50 wet:dry ratio): The severity of the frequency variations is about half (in terms of dB) of those found above in the BYPASS mode, yielding a smoother response.
Tested (wet): Same as above.
Signal-to-noise (@ nominal operating level):
Claimed: 88dB.
Tested: Unable to confirm. Given

a maximum usable output measurement of 10.5dBv and noise measurements on many patches below the -70dB measurement limit of the test equipment, the claimed figure is possible. All in all, we were impressed that most patches sounded extremely quiet.
Total Harmonic Distortion (BYPASS mode):
Claimed: Less than 0.08% THD @ 1kHz.
Tested: 0.028% THD @ 1kHz (+4dBv inputs and outputs) with INPUT control adjusted to light 6dB headroom indicator; 0.12% THD @ 1kHz (+4dBv inputs and outputs) with INPUT control adjusted to light 0dB headroom indicator. While this figure exceeds the maker's claim, it remains an inaudible amount of distortion in practical use.
Maximum input level (BYPASS mode):
Claimed: +18dBv (ref 0.775dBv)
Tested: +11.5dBv for 0dB headroom indication with INPUT control to minimum.
Maximum output level (BYPASS mode)
Claimed: +18dBv
Tested: +12.5dB absolute maximum signal output with OUTPUT control to maximum, INPUT control raised until onset of excessive clipping; +10.5dB maximum output to meet THD specifications. These level shortcomings will not affect most users.

recording studios, since the wet:dry ratio can be adjusted at the mixing console. It may be impossible, however, in the personal studio that may lack effects sends and returns.

In other bench tests, we received generally good results—although again, most fell short of the manufacturer's claimed specs. With the MIX control set to WET, the top-end frequency response never exceeded 17.5kHz. Distortion specs were fine, as were signal-to-noise specs. However, we weren't able to achieve claimed results with either of these.

Conclusions. The DSP256 does a great job at what it's designed to do—provide high-quality multiple effects at a moderate price. The remote is a nice touch, although some kind of rubber pad on its back would prevent it from moving around on a smooth surface. The unit sounds good, has a user interface that's easy to deal with, and integrates well into any MIDI environment.

While we did encounter shortcomings in the test results, we stress the following: In this intensely competitive price range, *most multi-effects devices have optimistically rated specs*; and realistically, all the problems we encountered are surmountable or probably inaudible.

At $529.95—including the remote—the DSP256 is a bargain. We recommend the DSP256, as long as you follow our suggestions to maximize its performance.

—*Dave Bertovic & Brent Hurtig*

Rane DC24 Dynamic Controller

THE PEOPLE AT Rane are to be admired for their slightly eccentric approach to building audio gear. Many of their products come with a twist—an equalizer with a built-in spectrum analyzer, for instance, or a 6-channel power amp that can be configured as six mono amps, three stereo amps, two stereo amps + two mono amps, and so forth. This eccentricity also extends to their sense of humor. (Last year a new product sheet for the Rane Psychoacoustic Infector circulated the industry; this brilliant—and fictional—bit of engineering sported a non-resonant concrete chassis, happy-face XLR connectors, and a telephone jack.)
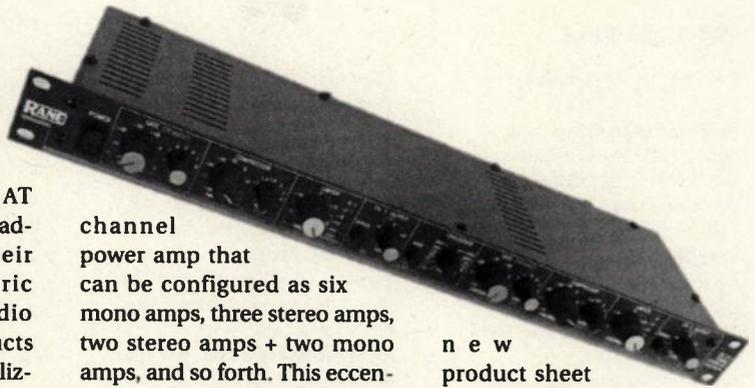
The DC24 is built in the Rane tradition. It is a dual-channel dynamic controller with compression, limiting, and expansion/gating. The ability to use all three of these functions simultaneously is impressive (most other dynamic controllers offer such functions just one or two at a time). Of course, there's a unique twist: a built-in crossover. And the manual—while its operational descriptions are a bit skimpy—is a great read. The 6-page guide begins with, "Shredded, this document makes excellent packing material." All this unbridled creativity must have something to do with the weather up in Washington state.

Overview. The DC24 stands one rack-space high, and measures just over 5" in depth. This shallow depth is a result of the unit's use of an external AC power transformer (it is hard to reach the rear panel, however, when the unit is racked between two much deeper boxes). The power supply sends 18 volts AC to the DC24 through a thin cable similar to a modular North American telephone cord.

EQ Background Notes

Dynamic Processing

Dynamic processing is almost essential in the modern studio. Basically, it is the control of signal level. Almost all dynamic processors use VCAs—*voltage controlled amplifiers*—to control levels electronically.

Compression reduces the overall dynamic range of the signal, making loud passages of music seem quieter and quiet passages seem louder. This is most useful for reducing excessive level variations in vocal and instrument tracks. Compression also can be used to increase the apparent sustain of an instrument, such as a bass, since the signal gets boosted as the instrument's sound starts to decay. (Hiss and other quiet background noise also get boosted.)

Limiting is like an extreme form of compression, although normally it's designed just to put a "cap" on how much signal will pass through. If you have signals that are peaking out of control, such as the raspy screams of a heavy metal vocalist, it may be time to use the limiter.

Expanders are the opposite of compressors: They *increase* the overall dynamic range of signals, making quiet passages seem quieter, and louder ones seem louder. Since tape hiss is relatively quiet, an expander can function as a single-end noise reduction system. It's also useful for quieting unwanted "tails" of a sound, such as the ringing of a just-struck drum. Gates are an extreme form of expansion. They are used to completely silence signals. The now-classic (and overused) Phil Collins/Hugh Padgham gated drum sound is created by allowing drum sounds to reverberate for a moment, then cutting them off instantly with a gate.

All these dynamic processing functions start to do their magic once signals cross the THRESHOLD level. Below this level (in the case of compressors and limiters), or above this level (with expand-ers and gates), the processor does not affect the signal.

Many processors—the DC24 included—have an external side-chain input. Normally, each channel's VCA receives its signal from the channel input, so the VCA is controlled directly by the level of the input. As we describe in our review, however, it's possible to use an external signal (such as that from an EQ) to control the processing of the VCA. We'll explore some side-chain tricks in an upcoming issue of *EQ*.

—*Brent Hurtig*

If you're grumbling about yet another external power supply to add to your studio, don't—this one makes sense, for several reasons. First, conventional power cords carry 120 to 240 volts; 18 volts is much less likely to induce buzzing and other noise as it passes by audio cabling. Second, a conventionally powered device must, by law, have its internal circuitry and chassis grounded to AC power ground. This can induce the dreaded *ground loop*, a common source of buzzing. The DC24, thanks to the external supply, is allowed to have a switch that "lifts" the chassis ground from the power supply. Our testing confirmed this switch's ability to reduce noise in some wiring situations. Third, the power transformer is located outside the steel chassis, further reducing noise. Fourth, it costs manufacturers big bucks to get UL and CSA power approval; with an external supply, one approval works for multiple boxes, and could save consumers money. Finally, Rane and several other companies are trying to make this power supply scheme a standard for rack-mount gear. In time, you may be able to power all your rack gear from a single rackmount power supply, and enjoy the aforementioned benefits.

With a flick of a switch, either channel of the DC24 will operate at "pro" +4dBm levels, or at "semi-pro" -10dBV levels. The audio input/output (I/O) connectors are 3-pin balanced XLR. We think a better choice would be 3-conductor (balanced) 1/4" jacks, because they are acceptable to many commercial users, and don't force small studio owners to rewire cables with XLRs. Still, the DC24 has many applications in live sound, where sound engineers are simply crazy about locking XLRs. The side-chain I/Os are 2-conductor (unbalanced) 1/4" jacks.

Operation. The DC24's two identical channels can operate independently or "slaved" together. (Well, they're almost identical—when the crossover is engaged they receive different signals, as we'll learn.) The expander/gate and compressor sections each have their own separate THRESHOLD and RATIO knobs; the limiter section has a single THRESHOLD control. Each side also has a GAIN REDUCTION meter, showing the amount of processing being performed, as well as an OUTPUT LEVEL control to compensate for loss of level caused during compression or limiting.

The BYPASS button for each channel created a tiny bit of clicking noise when pressed while a signal was present, even after a number of pushes to clean it out. The bypass is a "hard wire" design, which lets the signal pass through the box even if power is lost. As described below, we found this to be not always the case.

As a compressor, limiter, or expander/gate, the DC24 is easy to use and works wonderfully. With the limiter section, for example, just set THRESHOLD to the level at which you want limiting to take place. Once set, it's virtually impossible to exceed that level. The compressor section also kicks in when signals exceed the threshold setting; the RATIO control lets you adjust how *much* compression occurs. With as much as 6dB worth of gain reduction, we were pleased not to hear the effects of compression. Beyond 12dB, however, we were definitely in the land of the Big Squash.

When used with a mild-to-moderate RATIO, the expander/gate functions like an expander—increasing the overall dynamic range between loud and soft passages. With the RATIO control set all the way to "10:1," severe gating takes place: Signals that fall below the threshold setting are effectively silenced.

Many compressor/limiter/gates include ATTACK and RELEASE controls. These determine how much time it takes for the processing to start or stop once the input signal has crossed the threshold level. The DC24 has no such controls. Rather, it has automatic, or "program-dependent" attack and release times. Essentially, signals that *gradually* increase or decrease in level receive slower attack/release times; quick signals (like a snare) receive quick times.

Normally, this approach works fine; no muss, no fuss, and two less controls to worry about. But once in a while these controls are missed: when you want to hold open a gate just a bit longer to let a drum resonate, for example, or make the onset of an expander a bit more subtle, and so forth. The good news is that generally you can adjust the THRESHOLD control to perform these occasional tricks.

The DC24's side-chain input worked flawlessly. Here's one of the tests. With a microphone signal in channel 1 of the DC24, we patched the SIDE CHAIN SEND to an EQ. We then boosted the mid- to high-bands of the EQ and routed them back into the SIDE CHAIN INPUT. The result? A frequency-dependent compressor known as a *de-esser.* The compressor section of the DC24 compressed the mid- to high-frequencies of the voice before it compressed the others, resulting in less overall sibilance.

Why a crossover? The crossover's most obvious application is for "biamping" a P.A. system. On the back of the DC24 is a SEPARATE/COMBINE OUTPUT switch. When SEPARATE is selected and the crossover engaged, signals coming into the channel 1 input are split by the crossover into two frequency bands. The crossover range is adjustable from 70Hz to 7kHz. These two bands, low

and high, then can be processed by each of the two channels of the DC24: Low frequencies receive one compressor/limiter/expander/gate setting, highs can receive another. The processed bands show up separately at the channel 1 and channel 2 outputs. These signals then can be patched to two different power amps (or amp channels) that in turn drive low- and high-frequency speakers. Clever.

In the studio, the crossover has some different applications. With COMBINE selected and the crossover engaged, a signal entering channel 1 is split into two bands. These two bands again may receive separate processing. What's different here, though, is that the two bands' signals are merged at the channel 2 output. Why bother? This little exercise allows you, for instance, to apply different amounts of compression and limiting to the low and high ends of a piano. Or let's say the saxophonist sounds great, but every time she hits the high C she pins the meters: Just the high-end of the sax could be limited. *Very* clever.

You also can use the SEPARATE mode in the studio. With this setting, the crossover acts like a low-pass filter to signals in channel 1, and like a high-pass filter to signals in channel 2. We found some great-sounding guitar, vocal, and keyboard tones using the DC24 in this equalizer-like manner.

We did find a quirk that we didn't like. In COMBINE mode, if channel 2 is bypassed, all signals to the channel 2 output are killed. This happens because the output is looking for the non-existing input to channel 2. Unfortunately, it also prevents you from comparing a dry high-band signal with a processed high-band signal.

Conclusions. The Rane DC24 sounds good, performed well in our lab test, and is easy to use. Some people may miss ATTACK and RELEASE controls; others may be bothered by the choice of XLR connectors and the COMBINE mode bypass problem. Minor quibbles. If you're looking for a dynamic controller, the DC24 is the best built, most versatile, and most professionally equipped device you'll find for the money.

—*Brent Hurtig*

Korg T-Series Music Workstations

KORG'S NEW T1, T2, and T3 music "workstations" improve on the concepts initiated in the M1, last year's most popular keyboard-based product. Like the M1, these machines combine a large complement of excellent 16-bit PCM instrument and drum sounds, two stereo digital signal processors, four audio outputs, and an on-board 8-track MIDI sequencer for simultaneous use in composition and performance.

The three different models are largely distinguished by their keyboards. The T1 has an 88-note weighted action , the T2 features a 76-note synth-type keyboard, and the T3 offers a similar clavier with 61 keys. The other difference is the internal memory structure—the T1 comes standard with an additional 512k of RAM for additional PCM sounds, which is op-

tional on the T2 and T3.

Programs. The T-series doubles the M1's storage capacity with 200 user-programmable locations for sound PROGRAMS. As with the M1, a PROGRAM can have one of three architectures—SINGLE, DUAL, or DRUM. SINGLE uses a lone oscillator, filter, and amplifier, along with a healthy complement of envelopes and modulators, to achieve 16-voice polyphony. Each oscillator can draw from any one of the 190 PCM waveforms available between the original M1 ROM and the new T1 ROM.

DUAL mode essentially pairs two complete SINGLE sound-generating chains in parallel for 8-voice polyphony. OSC 1 parameters now can be copied to OSC 2 with a single menu command. DRUM mode allows any of the 85 PCM drum sounds to be mapped to any of 60 keys, doubling the M1's capacity.

Each PROGRAM now can recall its own unique velocity curves, keyboard temperament, aftertouch curves, and foot controllers. Multiple T- and M-series products also can be strung together as one big polyphonic instrument, given the new MIDI OVERFLOW mode.

Combinations. Like the M1, COMBINATIONS consist of up to eight simultaneous PROGRAMS with appropriately reduced polyphony. In addition to its own level, transposition, and keyboard range, each PROGRAM in a COMBINATION now can have its own pressure and velocity curves, MIDI controller assign-

ments, and keyboard scales. COMBINATIONS also can control external MIDI devices on up to eight simultaneous MIDI channels, including associated PROGRAM CHANGE commands.

Sequencer. The on-board 8-track sequencer retains the same basic linear (rather than pattern-oriented) song architecture as the M1. While patterns can be recorded in loop mode *à la* drum machine programming, they unfortunately still only can handle a single track at a time (i.e., you can't record a bass pattern while listening to the drum pattern destined to go with it). Editing functions such as quantizing, copying, inserting, and deleting measures also can be performed only on a per-track basis.

We also found it quirky that changes to sequence parameters made during playback (such as tempo or sounds) would revert automatically to their former status when the sequence was stopped, unless the WRITE function was invoked. All in all, you can realize most compositional visions in SEQUENCE mode, but getting there isn't much fun.

This is not to say that the sequencer has not been improved over that of the M1. Sequence capacity has been increased to 20 songs of up to 999 measures each, and 200 patterns can be defined of up to 99 measures each.

Alternating time signatures now are supported. Notes can be edited audibly while scrolling through the event list.

Changes in tempo, program parameters, and volumes while the sequencer is running now can be recorded. Each song can have its own settings for foot controllers, aftertouch, and velocity curves. Controller data is updated automatically, so starting playback anywhere in the song doesn't leave a performance parameter, such as pitch bend, stranded on Mars.

Effects. The T-series also inherited two independent stereo digital signal processors including reverbs of all sizes, delays, chorus, distortion, and equalization. Effect routing can be parallel or serial. Four outputs are provided for the effects and/or dry panning. While a few of the effects exhibited audible artifacts, the processing is as good as any you'll find built into an instrument.

The effects types and settings for both processors can be programmed for each PROGRAM, COMBINATION, or SEQUENCE. The effects become such an integral part of the sonic character for each PROGRAM that we confess to being spoiled; we missed the ability to retain the individual effects for each PROGRAM *within* a COMBINATION or SEQUENCE. A quick reality check reminded us that two years ago, on-board DSP didn't exist at all!

LCD Display. One big improvement is the enlarged

Description: Keyboard-based MIDI workstation.
Manufacturer: Korg USA, 89 Frost St., Westbury, NY 11590; (516) 333-9100.
Price: T1—$5669.00; T2—$4169.00; T3 - $3749.00; additional RAM—$250.00 installed.
Selected Specs: 16-bit PCM sounds, maximum 16-voice polyphony, 8-track MIDI sequencer, two programmable stereo DSP effects.

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graphic display and the user-interface enhancements it makes possible. For example, when you select a PROGRAM or COMBINATION bank prefix, a menu pops up presenting the ten selections available within it. The basic PROGRAM display has room to show some of the main parameter attributes, such as which effects are in use. Envelopes and keyboard ranges are displayed graphically as well as statistically. The various PROGRAM levels within a COMBINATION are displayed intuitively as bar graphs, as well.

RAM Option. The additional RAM found in the T1 (optional on the T2 and T3) provides space for additional 16-bit PCM sounds that can be loaded from floppy disk or via MIDI Sample Dump. In the latter case, each individual sample is loaded one at a time and takes its place as an additional DRUM sound; the user can construct a playable multi-sound instrument after importing and mapping the desired samples.

Disk Drive. High-density floppy disks are used to archive any type of T-series data. Disk capacity is four sets of the complete memory contents except PCM RAM data, or one set of all data including PCM RAM. Unfortunately, files can be named only "A," "B," "C" or "D," so manual documentation becomes crucial. T-series drives also can read Korg DSM sampler files or be used to archive up to 64k of SysEx bulk data dumps from other gear attached to the instruments' MIDI ports.

Conclusions. The T1, T2, and T3 are incredible as instruments, controllers, and MIDI sound sources. Both the electronic and weighted actions feel good and respond well for their respective designs. While the sequencer is relatively powerful as built-in hardware

goes, that power is difficult to access if you're used to most software-based equivalents. Perhaps the ultimate compliment we can pay the T-series products is this: We had a hard time actually getting this review into print because the sounds are so inspiring, we found ourselves writing songs rather than words!

—*Jeff Burger*

Crown SASS-P PZM Stereo Microphone

STEREO MICROPHONES enjoy a strong following in the film industry. They are much less widely used in music recording, however—most cost well over $2,000, and a good stereo recording can be performed with a less expensive pair of cardioid mikes.

Nevertheless, the current popularity of digital sampling introduces a new marketplace for stereo mikes. Ever hungry for tasty new sounds and effects, droves of engineers now take portable DAT recorders out into the field to go where no recordist has ever gone before.

For most of these sample-holics, the cost and relative fragility of stereo mikes make them undesirable for field recording. Enter the new Crown SASS-P stereo microphone, both sturdy and quite affordable, and as competent in the field as it is in the studio.

Not Just Another Pretty Mike. The SASS-P doesn't look

anything like a traditional mike, stereo or otherwise. The arrow-head shaped box houses two PZM capsules mounted on boundaries, separated by a foam barrier. This barrier is key to the design: It allows each capsule to function as a directional mike. The relationship between the barrier and the capsules creates a near-coincident type of pattern that is fully mono-compatible.

By proportioning the microphone structure to roughly that of a human head, Crown claims to have created a system that closely simulates the way our own ears perceive audio imaging.

The complete package is quite nice. The system comes housed in a Haliburton-style, foam-filled aluminum road case. Included are a hand grip, two windscreens of different densities, and a thread adaptor to accommodate different stands. The owner's manual is a bit skimpy, with just four pages of specs and application notes; some placement diagrams and a bit more background on PZM miking in general would have been appreciated.

PZM mikes use electret condenser elements that require a power supply. In the case of the SASS-P, power can be supplied either internally via two 9V batteries or externally via a 12V to 48V phantom power supply (as found in most professional mixing consoles).

In The Field. To begin testing the SASS-P, I took it and a DAT recorder on location to record horse sound effects for the cable TV series, *Zorro.*
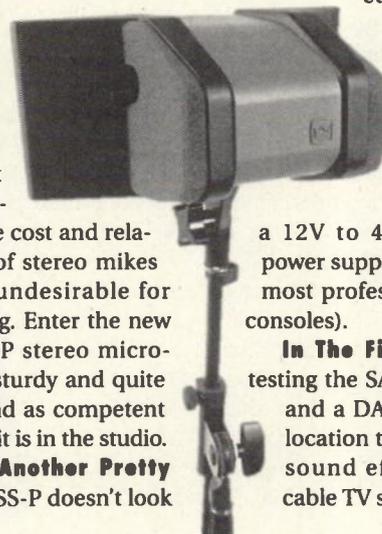
I engaged the mike's internal low-cut filters to reduce wind noise; these in combination with the windscreen were quite effective. I tried both "bys" (where the mike remained stationary as the horses passed by) and "steadys" (where the mike followed the horses as they approached and passed). I created another horse steady by giving the mike to the rider. In all cases, the imaging was wonderful—an extremely believable stereo image. The quality of the sound was smooth and ambient—there wasn't much of an "in-your-face" sort of sound.

Most stereo mikes suffer from an inordinate amount of hand-hold bumps and other noise, even with good shock mounts. The unique structure of the SASS-P virtually eliminated hand noise.

Exterior ambience recording was quite good. Again, the imaging was wonderful. This sort of low-level recording did reveal that the SASS-P suffers from a slight amount of internal hiss—something we didn't hear when we plugged in much more expensive stereo and mono mikes. The noise is somewhat more intrusive when the mike is powered by internal 9V batteries. The inter-

nal noise is virtually inaudible when recording higher-level stuff, such as horses or music.

Musical Applications. The SASS-P pulled off a really nice studio recording of a grand piano from a distance of about 5' from the propped lid. Close-miking the piano just above the hammers created a delightful subtle movement of sound when different ranges of the piano were played, though ambient room miking gave better results overall. Nice results were also achieved with an acoustic guitar; 4' away, above the neck, worked best for me.

A live rock band recording was detailed, although the club setting revealed the characteristic that is the mike's best and worst feature: It picks up lots of ambience (in this case, a noisy audience). This ambient sensitivity may be problematic in similar distant-miking settings—yet may turn out to be a plus in an environment such as a quiet concert hall.

Conclusions. While the mike is easy enough to handle, it looks big and bulky. I did notice, however, that few people recognized it to be a mike, which might have some clandestine benefits such as during the live TV broadcasting.

Describing a mike's tonal quality is as subjective as describing the sound of a concert piano. If you're familiar with mono PZM mikes, you won't be surprised by the SASS-P's sound. By injecting a bit of a boost in the 4kHz and 10kHz ranges at the mixing console, SASS-P tracks could be made to sound more up-front and defined, without changing the image or degrading the sense of ambience. Like other PZMs, it's virtually impossible to overload this mike.

The SASS-P is not the only stereo mike to offer great imaging and sound. And if your *en-*

EQ Background Notes

Stereo Miking

X-Y Miking. Stereo miking is fairly straightforward, and it's something anyone can perform with just a pair of mikes. Here's one technique known as *X-Y* or *near-coincident pair* stereo miking:

1) Take two cardioid (unidirectional) microphones.

2) Place them on two separate stands (or on one stereo mike holder).

3) Point their capsules together at roughly a 90°-130° angle.

4) Aim the v-shaped pair towards whatever it is you want to record.

5) Pan their respective mixer channels hard LEFT and RIGHT.

And *voilà*, a stereo microphone system. Straightforward? Yes. Convenient? No—especially not for field recording, where a hand-held double-mike system can be most unwieldy.

M-S: Two Mikes In One. Fortunately, stereo microphones are available from many manufacturers including AKG, Neumann, Sanken, AMS, Fostex, and Sony. Many of these mikes feature an *M-S* design, which stands for *mid-side*. M-S mikes typically have a single unidirectional capsule that points forward, along with a bidirectional capsule that receives sound from both left and right sides of the mike.

Most M-S mikes come with an external *decoder* box that lets you vary the "width" of the stereo field; that is, by tweaking a

knob, you can vary the stereo panorama all the way from mono (the same information in both channels) to a broad stereo image. Extremely wide settings are sometimes not "mono compatible," since phase cancellations may cause the loss of information when the signal is heard in monaural through a TV or radio speaker, for example.

PZM Stereo. For some years now, the folks at Crown have been making variations of the pressure zone microphone (PZM). Most PZMs consist of a small omnidirectional capsule mounted within a millimeter or two of a flat surface. This surface is known as a *boundary*; other manufacturers refer to their PZM-style mikes as *boundary* mikes. Whatever they're called, such mikes are well-liked by many engineers for their exceptionally flat response and ability to withstand high sound-pressure levels.

Crown does have a series of mikes that use unidirectional PZM capsules, but most PZMs use omnidirectional capsules. Crown formerly made unidirectional PZMs with omni capsules by using physical barriers to reject sounds that emanate from behind and to the sides of the mike. It's this same sort of barrier technology that makes PZM stereo miking possible: The new Crown SASS-P has two omni PZMs, separated and made directional by a foam barrier and the boundaries.

—Brent Hurtig

tire mike budget is less than a grand, a pair of good cardioid condensers (such as AKG 460s) and a stereo mike clamp may provide more versatility than the SASS-P. Still, the SASS-P is an excellent value when compared to other stereo mikes in the $2,000 to $5,000 price

range, and it offers fine spatial placement. If you're in the market for a stereo mike—at any price—be sure to audition the SASS-P.

—Scott Gershin

"Badman's Song," which I think we wouldn't have been able to do on analog because we started off with the drums and had about 30 drum takes and we would pick the best bits. Then we would do the same for the bass, independently of whether it was played at the same time as the drums. When you're cutting analog, everything has to go at once.

EQ: *Some elements you've obviously borrowed from the Beatles, especially in the title cut, "Sowing The Seeds Of Love." What do you learn when you set out to do that?*

Curt: Another way of doing something. The idea was to kind of make it a Beatles pastiche, the reasons being musical because the lyrics are politically serious and we didn't want to be labeled as some opinionated left-wing kind of young people, so we decided to couch it in this kind of humorous backing track. The song's chorus conjured up images of the late '60s, so musically we wanted to do it that way. We wanted to add a bit of humor. So we thought, well, why don't we just do this Beatles pastiche and make it fun to do?

What you do learn is that sometimes simplicity really works, in the sense of trying to build a song just using real musicians and real music. We didn't sequence stuff on that track. In the first verse you've got a little bit of drums and a bass and Wurlitzer piano and that's it. And strange, I mean very highly compressed, John Lennon-type vocals. Then we get to the big chorus. It was really good fun to do. We had a whole orchestra in for a while to do all the

Producer's Tip:

"One of my favorite guitar bits on the new album is just a piece of feedback on the end of "Woman In Chains" just before the tag. The guitar I have is a PRS—Paul Reid Smith—and the bridge and tremolo arm are so good that you can totally depress the arm. Strings go completely floppy and you can bring it up from nothing to a chord. That's how that feedback came about. We were monitoring very, very loud and I had the bridge deflated, brought it up and it fed back. It's wonderful. We're painstaking sometimes in the double-tracking, like with the arpeggio sound on the chorus of "Sowing The Seeds Of Love." It's four guitars, completely different guitars and setups, from DI's to Roland GP-8 modules to a very tiny Boss amp and speaker, like a tune-up amp."

—*Roland Orzabal*

cello bits and little bits of orchestra on the song, and that was also fun.

EQ: *Nothing was actually sampled from Beatles records, right?*

Curt: No, me and Roland don't even *have* any Beatles records.

EQ: *Did you try things like reverse tape loops and varying the speeds of the voices?*

Curt: Yeah, we did a whole bunch of stuff. We got a little Vocoder and made it kind of spacey and played around with it. It was intended to be humorous and it was done with a lot of love. We really enjoyed it. It came very easily. The whole thing was, it was done from memory. I think it would have been wrong for us to sit down and listen to Beatles records and think, "How did they do that?" We just picked out the bits that were in the back of our heads from listening to them over the years at other people's houses or on the radio. The things we always remembered were the Wurlitzer piano, highly compressed vocals, spacey Vocoder here and there, and cellos, orchestra, trumpets.

EQ: *Do you use some favorite effects throughout the album?*

Roland: For basic reverb we use a Yamaha Rev7, the large hall sound with a bit of EQ. Then in terms of mixing, a plethora of reverbs came out, like the new Lexicons and all that kind of stuff. Pretty much the same as everyone uses.

EQ: *Let's talk about special guitar effects. You have things that sound very '70s-like, such as wah wah-type guitars. Foot pedals?*

Roland: I used what's called a "rat pedal" [Pro Co Rat] for lead guitar. I generally use the Boss compression sustainer, the blue pedal. And I use the Roland JC-120 stereo chorus amp. I have quite a few guitars, and a Marshall amp, and I recently acquired a Soldano, which is wonderful. I saw Knopfler and Clapton at the Albert Hall using one.

EQ: *How did you do the sound effects, like the rainstorm and crowd noise on "Year Of The Knife"?*

Curt: Some effects came off BBC sound effects tapes. The rain was actually the crowd noise from one of our live concerts. It was intended to be the album's opening track, but we decided it didn't work in that sequence. We wanted it to be like a big kick-in. So we did the drum click—not a click track, but with the drummer hitting his sticks together—straight in, with the audience cheering. We had that on the tape.

> *"It would have been wrong for us to sit down and listen to Beatles records and think, 'How did they do that?'"*
>
> **—Curt Smith**

Then we thought we might leave it off since it wasn't going to be the first track, but it worked so well coming out of "Standing On The Corner [Of The Third World]" into "Swords And Knives" because those are mellow tracks. You suddenly get woken up again by this sort of crowd noise. So we left it there. We had one live cut on the last album, too. ["Head Over Heels" segues into a live "Broken" on Songs From The Big Chair.]

EQ: *Do you have a favorite vocal line or bass riff, and could you describe how you did it?*

Curt: On the last album I like "The Working Hour" because it had a very musical bass line that just came about in rehearsals. On "Sowing The Seeds," we're just trying to recreate kind of dull sounds with this old Fender semi-acoustic bass.

EQ: *Did you say dull?*

Curt: Yeah, dull, on an old Fender semi-acoustic bass that you just play with your thumb. Not slapping it, just playing it softly. You get one of those little old dull sounds you used to get on old records. It was nice.

EQ: *The new album liner notes don't identify which voice is whose. Since the first record you haven't identified your voices. Is there a point where your two voices mesh so indistinguishably that you feel, "We really put one over on them this time"?*

Curt: No. Most people get confused as to who sings what, but it seems blatantly obvious to me, which is why we never put the credits on. We have quite different voices, unless we're singing together, and then you can't really tell who it is because sometimes we do stuff in unison.

EQ: *I've never seen you play live, but I'm going to hazard a guess that Curt has the slightly higher voice.*

Curt: You got it. Mine's actually softer, a lot softer than Roland's. It's quieter. Roland has a very loud voice. I normally sing the stuff that's softer and a bit more melodic. If it requires any sort of vocal acrobatics, then that's Roland's voice.

EQ: *So it's you singing "Sowing The Seeds"?*

Curt: The chorus. Yeah.

EQ: *What percentage of things recorded at Roland's house actually made it into the final recording?*

Curt: About 60 to 70 percent.

EQ: *So you didn't do a whole lot of rerecording when you went to the bigger studios?*

Curt: No, mainly it was just to do the loud stuff, like cranking guitars, or to get the drums in, stuff we couldn't do at Roland's

house.

EQ: *The album cover doesn't give any studio credits—where did you do the mixing?*

Curt: We did a lot of it at Roland's house, 'cause we did have the one 32-track machine and we just slaved stuff off and did a quick mix down to two tracks, then worked on top of it and just kept the other 32 tracks. When we needed to put them both together, we used mainly three studios in London—Mayfair, Townhouse, and Olympic.

EQ: *What console did you use for final mixing?*

Roland: We mixed down at Mayfair Studios for the majority of it because they had a 72-channel SSL desk with a mixture of G series and E series electronics.

EQ: *You've been quoted as saying there was a calculated effort to make "Everybody Wants To Rule The World" [Songs From The Big Chair] into a commercial song. Was there a calculated effort to do so with any song from the new album?*

Curt: Not on this one, no. Not at all. That was nice to do on the last album. It was an exercise in musicianship, getting something to work that way, and it was what we wanted to do. This time we wanted to treat every song separately the way we felt it should be treated and make an album that was relatively diverse in tone. I think we succeeded in that. I don't think you get as bored with this one because it's got so much stuff. It has so much to give.

EQ: *Looking back on these sessions, have you reevaluated your whole studio approach? What's going to be different next time?*

Curt: The reevaluation is that there are no rules. The next time it will be whatever feels right at that time. There's just no point in planning it. Can't say what it will be right now 'cause I'll probably feel different by the end of the tour. You make all these grand plans the year ahead, then spend six or seven months on tour. You discover a lot about music when you're on tour, especially playing with new people. So when that's finished, my view of music will have changed again and I'll probably go on to do something different. •

DEVON BERNARDONI

finish the entire year-long project. It was a hard album for everybody, and it came out great. That's a wonderful feeling. And it was thrilling having the opportunity to work with the same band I listened to when I was a kid."

Cycles was cut at the Plant on a digital Mitsubishi X-850 32-track tape machine for most parts; an analog Otari MTR-90-II 24-track was slaved for occasional percussion and vocal overdubs. "I love the Mitsubishi, though given the choice, I still like the sound of analog for some parts, especially drums," notes Bernardoni.

In late 1989, she was back in the studio—co-engineering with Jim Gaines—on Carlos Santana's new album *Spirits Dancing In The Flesh*. Simultaneously, she engineered and co-produced Jeff Narell's Rhythm & Steel group. Future plans include a children's album (for which she's writing songs), a greater emphasis on production, and securing an agent: "Finding the time to represent myself these days is a real challenge."

Rising to challenges has helped Devon Bernardoni become a rising star in the engineering world—and has required her to be completely comfortable with the gear and techniques found in world-class studios. But for the transplanted Canadian, understanding the science of recording is absolutely secondary to the art: "Technology is something anybody can learn. Sure, it's important, since it's the tool we use to capture music. But ultimately, helping someone to *communicate* to other people—through their music—is what's really important." •

Producer's Tip:

"Good engineering captures the energy and feel of the music—and that applies whether you're working with a band or alone with a sequencer. To do this, preproduction and planning are essential. *Think ahead.* For instance, when I cut basic tracks, before the session I'll have already considered microphone choices and placements, and I'll have second choices ready to try, just in case. When a session really flows creatively, ideas develop because you gave them the opportunity to flow." •

SHARP ANGLE

Product developers should be required to spend more time in the field absorbing input from an array of end users...

new. Then product ideas must flow from a marriage of the dreams of designers with the hard, everyday realities of people who make their living using the equipment. Many products we perceive as brilliant are simply the obvious solution to an everyday problem. The solutions are there, waiting to be discovered by a fresh approach, clear mind, and open eyes, but can't be seen by those who aren't looking! Product developers should be required to spend more time in the field absorbing input from an array of end users, and less time at the office, where ideas are exchanged within the same closed circle of co-workers. An obvious point, oft-overlooked, is that in this business, the *sound* of a device is critical; it should be a crime to bring a product to market without first exposing it to a varied audience and listening to the comments on sonic performance.

Also, products are maimed when they are cost-controlled to death. The market research and engineering teams may have done their jobs, but management becomes paralyzed with fear when they realize that their new product might sell for 20 percent more than a less-powerful competitor. Again, courage is needed. If the "wish list" for a product is clear, the product simply should be built according to the list, and only *then* should a price tag be hung upon it. That way the end product is not a hopeless compromise between an end user's dream and an accountant's nightmare.

Professionals are power users. They are willing to spend more for a product with a meaningful advantage.

When they give more attention to the needs of their customers, and get the guts to go out on a limb once in a while, equipment manufacturers will see a higher percentage of their products become best-sellers—and end users will get the power tools they need. •

ADVERTISERS INDEX

COMING NEXT ISSUE

24-TRACK
SHOOTOUT

ARRANGING
IN THE STUDIO

STUDIO REFERENCE SERIES

It's very common to hear chords from the neighboring counter-clockwise key in pop, rock, gospel, bluegrass, and country music.

Every key has a hidden chord that's not shown on the wheel: It's a diminished (°, or *dim*) triad, located a major seventh away from the I chord. Counting this vii°, a key contains seven triads: I-ii-iii-IV-V-vi-vii°. The key of *G*, for instance, comprises *G* (I), *Am* (ii), *Bm* (iii), *C* (IV), *D* (V), *Em* (vi), and *F#°* (vii°).

Note: Each triad on the cycle chart can be extended diatonically. For example, played as four-note chords (adding the note which is a third above the top note in the triad), I and IV become major 7th chords; V becomes a dominant 7th; ii, iii, and vi become minor 7ths, and vii° becomes a half-diminished 7th.

Cycle Chart Applications. While noodling on a keyboard, let's say you stumble across a groove containing four chords—*Dm*, *Bb*, *C*, *F*. You then wonder, "What other chords can I use to expand this progression?" Locate *Dm*, *Bb*, *C*, *F* on the wheel: With a little effort, you'll see that they fall within the territory of *F* major. (They are the vi, IV, V, and I chords of the key.) Once you've nailed down these four chords, you can see that *Gm* and *Am* also belong to the key of *F* and are, therefore, potential candidates for the next part of your groove.

Here's a cool trick: Depending on your melody, you can effectively borrow chords from the two adjacent keys, *Bb* and *C*. (Notice how any two adjacent keys always share four chords.) It's very common to hear chords from the neighboring counter-clockwise key in pop, rock, gospel, bluegrass, and country music.

The cycle chart makes transposing a breeze. Write down the chord names of the progression you need to transpose. With your finger, trace the movement from one chord to the next on the wheel. Pick a new keynote, and recreate this motion exactly. Presto—instant transposition!

—*Andy Widders-Ellis*

ANNOUNCING!
EQ
CLASSIFIED ADS

GOT SOMETHING TO SELL?

Your Classified ad in *EQ* will reach thousands of recording industry professionals and musicians.

Create the *right* image for *you*: Choose from two formats: a Regular Classified, featuring several options (borders, screens) to make your ad stand out, or opt for a Display Classified to show off your logo or artwork. Either way, your ad will be seen by the people you want to reach.

Companies *and* individuals are welcome to advertise in the Classifieds. Companies, of course, want to reach as many savvy customers as they can. But individuals also find Classifieds a great way to sell used equipment.

CATEGORIES AVAILABLE

Please mark your ad with the category under which you wish it to appear. If you don't choose one, we'll put your ad in the most appropriate category. Choose from:

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

REGULAR CLASSIFIEDS

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

RATES

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

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Prepay for four issues, get the fifth **free.**

MODEL ADS

This is an example of a Regular Classified ad with no extras. Simply count the number of lines, figuring 30 characters per line. This ad contains 6 lines. At $10.00 per line, it costs $60.00.

This is a Regular Classified
with a black border
and a few lines in bold.
Lines may be centered, if you wish, for effect. Once again, it's $10.00 per line.
Add $10.00 per bold line,
and $10.00 for the border.
10 lines x $10 + $30 + $10 = $140.00.

This ad has a gray screen and one bold line. All screen ads have a border (no extra charge). It contains five lines. 5 x $10 + $10 (screen) + $10 (bold) = $70.

This ad has a knockout (white print on black background). It has four lines. 4 x $10 + 10 (knockout) = $50.00.

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

RATES

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

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

MORE MODEL ADS

This ad is 1" high, and costs $80.00. If you can say it in a small format, this ad is for you!

EQ CLASSIFIEDS

This is a Display Classified **S**nappy **s**logan! that is 1-1/2" high. It costs $120.00.

Sloppy Artwork!

Need to list lots of items?

This 2" ad holds a lot of information, at a nice price: $160.00.

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Vi Very impressive, Inc.
P.O. Box 4032
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DEADLINES

May/June issue: **March 1**
July/August issue: **May 1**
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Nov/Dec issue: **September 1**
Jan/Feb issue: **November 1**
March/April issue: **January 1**

THE FINE PRINT

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

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

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

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

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

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

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

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

NEED TO KNOW EVEN MORE?

Write:
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Cupertino, CA 95015

FAX:
(408) 446-1088.

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

The Side Effects Of Technology

HIGH-TECH SOLUTIONS SOLVE high-tech problems—most of the time. More often than not, it seems high tech just digs the hole deeper. Console automation makes you spend more time in the studio, and disk-based audio editing takes twice as long as conventional digital audio editing, which takes twice as long as razor-blade editing. The list is endless. At least once a day I find myself uttering what has become my current motto, "It's always *something!*" It seems as if nothing ever does what it is supposed to do, or as fast as it is supposed to do it. In this first column I am going to grind my digital multitrack axe.

Last summer I had the opportunity to use the new Sony PCM-3348, the first 48-track digital multitrack recorder. What a nice machine. All 48 tracks are stored on 1/2" digital tape that travels at 30 ips. It is a DASH-format machine like its 24-track predecessor, the PCM-3324. [*Ed. Note: The DASH format for multitrack stationary-head digital recording also is used in machines by Studer and Tascam.*] It also will play and record tapes previously recorded on the 24-track version. Cool! The 2x over-sampling convertors on the record and playback sides really warm up the high end. I think it sounds very close to the audio quality of the old 3M digital 32-track, which always has been my favorite-sounding digital machine.

Electronic editing between two PCM-3348 machines is a piece o'cake. You just park the two machines at the edit points, or hit an EDIT button on the fly, and bingo: The edit points and offsets are entered into the machines. Press the PREVIEW button if you want to hear the results before you perform the actual edit, or just go for it and press the EDIT button. The edit points you select have a one-millisecond resolution. By the way, the PD digital machines (ProDigi format by Mitsubishi and Otari) have a resolution of one SMPTE frame, which is 30 milliseconds, while the 3M machine could be made to perform edits with a resolution of 320 microseconds, or less than 1/3 millisecond!

The new Sony machine also has adjustable crossfade times for punch-ins and edits. The adjustment, located on the remote, allows for a range from 1.3 to 341 milliseconds. Great for a quick transition on the rhythm tracks and a nice long crossfade on the orchestra. Try doing *that* with a razor blade.

But then I remember my motto, "It's always something." Well, one little drawback in the design of the machine lies in the way Sony implemented the built-in sample memory. The sample memory is mono. I don't know about you, but with 48 tracks waiting for something to be recorded on them, I am going to record everything in stereo, or quad, or whatever I can get away with. If you want to fly a part from one chorus to another, you have to do it one track at a time.

You can program the start point of the sample playback from memory, but you cannot program the record start point.

This means you either have to push the button manually to start the sampling process, or trigger the sampling process from an external click. If you wanted to move both halves of a piano from one place in the song to another, it would be pretty hard to maintain the same stereo image because of the probable phase shift between the two channels.

If the record point were programmable, however, you could get one half, put it where it belongs, get the other half, move it, and still maintain the proper stereo image. Hey, Sony, how about a software update? Here's a great little feature that I would really like to use. (Reality check: Getting even small changes out of a company this big is about as easy as pushing a car uphill with a rope. I'll keep you posted on this.)

As my 48-track project progressed, I realized I needed to use a tune that previously had been recorded on a 32-track Mitsubishi X-850. The thought of transferring the tape analog-style between the two digital machines went against everything in my gene pool. Keeping two digital machines around went against my budget.

Saved by the bell. The studio where I was working told me they had a new box, the Otari CB-503, that hooks onto the back of Otari's DTR-900 digital multitrack to allow digital domain transfers between PD and DASH machines, in both directions. I used it to transfer a tape that was recorded on a Mitsubishi X-850 to the Sony PCM-3348. It worked flawlessly. This little box actually does what it is supposed to do. Oh well, every once in a while something slips through. •

I'll be talking to you from "across the board" because the editors of EQ have given me free rein to disparage and applaud any and all facets of recording. If you have questions or concerns about recording techniques, please send them to me, in care of EQ, at 20085 Stevens Creek, Cupertino, CA 95014. I'll answer as many as I can.
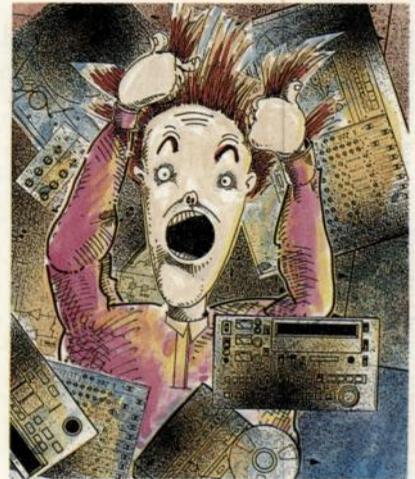
> "At least once a day I find myself uttering my current motto, "It's always *something!*"
>
> **BY ROGER NICHOLS**

ILLUSTRATION: SCOT HALPIN
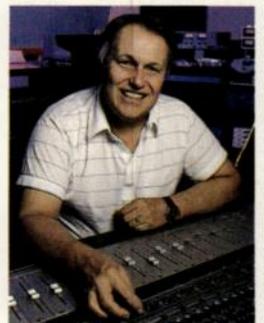
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T R I D E N T