Electronic Musician

November 1996

Modular Hard-Disk Recorders

7 magnificent machines strut their stuff

Create stunning piano samples

Get your demo heard

Check out the Tascam DA-38

Steal riffs from famous producers

U.S. $3.95/Canada $4.95

11

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1 VLZ CIRCUITRY FOR ULTRA-LOW NOISE AND CROSSTALK.

A fancy new name for the same old circuitry? Nope. VLZ (Very Low Impedance) is a Mackie innovation based on solid scientific principles. Through the careful deployment of high operating current and low resistor values at critical points in our consoles, thermal noise & crosstalk are dramatically reduced. Open up all the channels, subs and masters on an 8•Bus console and compare what you hear (or rather don't hear) with any Brand X console. And because VLZ circuitry needs loads of high current, we ship a humongous, 220-Watt power supply with every 8•Bus & 24•E expander.

2 IT EXPANDS ALONG WITH YOUR NEEDS AND BUDGET.

You'd be surprised just how many 8•Bus console setups like the one below are currently in use. But you don't have to start out this way. Start out with a 24•8 or 32•8 and then grow your 8•Bus console 24 channels at a time with our 24•E add-on modules. 1, 2 or even 3 of 'em connect in minutes. They come with their own 220-watt power supply; optional meter bridges are available.

3 IMPECCABLE MIC PREAMPS.

A console can have motorized dooflammers and an optional MIDI espresso attachment, but if the mic preamps aren't good, you don't have a fully-useful production board. Our discrete preamps with large-emitter-geometry transistors have won a critical acclaim for their exceptional headroom, low noise (−129.5dBm E.I.N.) & freedom from coloration. VLZ circuitry in the preamp section also reduces crosstalk.

4 THIS CONSOLE JUST PLAIN SOUNDS GOOD.

Sure, you may be able to buy a Brand X console for less. But you end up with a console that sounds like…well…a Brand X console. Granted, we're getting into a pretty subjective area here…but we have tall mounds of 8•Bus warranty cards that rave about our consoles' "clarity," "sonic purity," "sweet sound," "transparency," "lack of coloration" and a lot of other superlatives we wish we'd thought of first.

7 MAC & WINDOWS 95-BASED AUTOMATION THAT'S RELIABLE, PROVEN AND AFFORDABLE.

Along with affordable digital multi-track recorders, the Mackie 8•Bus has made it possible to do world-class productions on a modest budget. But until now, Big Studios have still had one remaining and unattainable creative "secret weapon"… computerized level automation. That's why we developed the UltraMix™ Universal Automation System. It gives you fully editable and recallable control of input, channel and master levels – plus features not found on even the most expensive proprietary Mega-Console automation systems. Equally important, it doesn't degrade sound quality, introduce zipper noise or cause audible "stepping." UltraMix is currently being used to mix network television music themes and on several major album projects – by seasoned engineers who grew up on Big Automation Systems. Their verdict is that UltraMix is a serious automation solution – stable, reliable and frankly easier to use than more expensive systems. The basic system controls 34 channels and can be expanded to as many as 128 channels. UltraMix Pro™ software, for 030/040 & Power PC Macintoshes and PCs (Windows® 95 required), includes a wealth of features like editable fader curves, built-in level display, up to eight subgroups, SMPTE time code display, event editor with pop-up faders, optional control of outboard effects devices, and the ability to play Standard MIDI files from within the program.

Above: 24•E 24-ch. expander with optional MB•E meter bridge and stand.

Above: 32•8 with optional MB•32 meter bridge and stand.

UltraMix™ includes the Ultra-34 Interface, UltraPilot Controller and software for $2797 suggested U.S. retail. Macintosh® or Windows® 95-compatible PC not included.

STUDIO *Quad*

GET 4 PROCESSORS

::: DigiTech

STUDIO *Quad*

4 IN 4 OUT MULTI-EFFECTS PROCESSOR

44 1:Hall 2:Rm 3:Dly 4:Dtn

4 COMPLETELY INDEPENDENT INPUTS

4 COMPLETELY INDEPENDENT OUTPUTS

4 COMPLETELY INDEPENDENT EFFECTS

*Buying one Studio Quad is like buying 4 independent processors. Inside the Studio Quad is a multi-effects processor that is multi-talented and while most of us can't do more than one thing at a time, the Studio Quad can do 4. Why did we put the power of 4 multi-effects processors into one unit? So you, too, would be able to do more than one thing at a time.

The Studio Quad features 4 completely independent inputs and outputs that give you the power of four independent mono processors or two true stereo processors. Imagine the power to process two true stereo sources simultaneously without sacrificing control. But true stereo is only the beginning. Imagine having 4 independent reverbs for vocals, guitar, keyboards, and drums, where each reverb can be optimized specif-

STUDIO *Twin*

SIMPLE. POWERFUL. AFFORDABLE.

You say you need a true stereo effects processor that's ideal for any application, a unit that is flexible enough to fit your unique situation, whether it is for sound reinforcement or studio applications. But it needs to fit your budget. Enter the Studio Twin Stereo Effect Processor. With 99 factory and 99 user presets, each with the 3 parameters you use the most at your fingertips, and a single band parametric EQ, it's **SIMPLE** to operate. The high performance, award winning SDISC® processing offers great sounding, high quality effects like True Stereo Halls, Cathedrals, Plates, Chambers, Rooms, Gated Reverbs single

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Do the math. It's your money.

ically to achieve the best audio performance for each application. And imagine a processor that gives you the power to create any combination of effects, in any order. And then multiply by four.

Check out the Studio Quad. With its ingenious user interface, large custom display, and potent S-DISC® technology at its heart, the Studio Quad is capable of not just replacing, but thoroughly eliminating the need for multiple, expensive, single-purpose processing units. You work hard for your money. DigiTech makes sure you get the most for it.

and multi-tap Delays, Chorus, Flanger, Detune,and Tremolo. **POWERFUL** Multi effects combinations like Mod+Delay, Mod+Reverb, or Delay+Reverb are also available. You can even process 2 independent mono sources through 2 separate effects. We also included a Noise Gate with an adjustable threshold, MIDI program change control, and an internal power supply. All these features packed into a single **AFFORDABLE** rackspace. Check out the DigiTech Studio Twin at you dealer today. Start sounding great tomorrow.

:::DigiTech

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I N S I

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REVIEWS

Big Is Better

These growing pains hurt so good.

You are holding in your hot little hands the largest issue of **EM** we have published to date. At 204 pages, this month's labor of love makes one heck of a plop when tossed onto a desk. I really dig that sound because it resonates with two echoes that are dear to my heart.

The achievement echo. I didn't have a personal vision when I became editor of this magazine—there is no avatar at **EM**; our "vision" is a collaborative effort by the entire staff—but I did have a personal *goal*. I wanted **EM** to become the pre-eminent magazine for the home recordist. I believe this month's mammoth issue symbolizes that we've slam dunked that objective.

For one thing, the November issue coincides with the Audio Engineering Society (AES) show—a convention that serves the recording industry—so it means a lot that many manufacturers supported this edition with their advertising dollars. Obviously, they share our belief that **EM** is a prime vessel to reach the burgeoning personal-studio community. In addition, we have consistently published larger issues than projected for this year, and our average page count rose to a whopping 172 pages! (When I joined the magazine as an assistant editor five years ago, we were publishing an average of 120 pages per issue.)

However, increased ad revenues do not just line our pockets with bucks. More ads mean we can run more pages, and more pages mean we can treat readers to more comprehensive features, more reviews, more eye-catching color layouts and photos, more gear face-offs, and more tips for a successful career. Everybody wins!

The teamwork echo. The continuing **EM** success story can be attributed to one thing: the dedication of its staff. Each and every one of them has sacrificed bits of their evenings, weekends, and even their vacations to the expanding needs of the magazine. Nobody works 9 to 5 here. *Nobody.*

Now, all of us have well-meaning friends who shake their heads and say that our work ethic is goofy, that only geeks and idiots would give up leisure time to toil, uncompensated, for the extra hours it takes to produce **EM** to our exacting standards. That's a fair knock. But I prefer to bow low and long to a staff that feels so passionately about its work that it refuses to deliver anything less than the best—even if it means that someone misses *Seinfeld* because a story could be slightly improved by a few incisive rewrites. It's truly an exceptional group of people I have the pleasure of working with, and their sweat and commitment is what makes **EM** an exceptional gear mag. (I'd also like to thank our wonderful freelance editors and artists—Mary Daly, Sally Engelfried, and Ingrid Wynden—for helping us survive the increased workload of producing 204 pages.)

Of course, meeting new challenges in 1997 will be so far from a piece of cake that it should be considered a bag of spiced liver chips. (Yum!) We'll sweat bullets thinking of ways to further improve the magazine. So, if you have any ideas about stuff that could make **EM** a better resource for your needs, please e-mail them to emeditor@aol.com. We'll probably stay up nights worrying over every little criticism and suggestion, but that's how we keep our edge. And don't be surprised if, next November, you toss a 220-page issue onto your desk.

Michael Molenda

I did it on an ASR·10.

Sony introduces the mini stu a better idea wh

From the people who created the MiniDisc format comes the home studio that makes it easy to keep creating. The Sony MDM-X4 Recorder.

Nailing the hook is as hard as ever; capturing it, however, just got easier.

Here's a machine that can keep up with your ideas.

It uses a digital optical disc which is read by a laser, just like a CD.

So now you can edit with instant access to any track without the waiting that comes with cassette fast forward and rewind. And it's easy to use, with a jog shuttle knob that helps you find what you're looking for faster.

How does incredible recording and mixing flexibility sound?

With MixWrite you can mix 4 tracks down to 2 "virtual" tracks on the same disc. With the X4's sophisticated song and track editing functions, you can record and

Your search is over: the new Sony MDM-X4 4-track recorder finds edit spots instantly, among other things.

combine 8, 12 or more tracks, creating complex productions that go far beyond what is possible with traditional cassette-based mini studios.
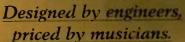
dio for musicians who know en they hear one.

And with MIDI In/Out/Through, the X4 can act as either a master or slave to MIDI. With MIDI Machine Control (MMC), you can actually operate multiple X4s from your MIDI sequencer.

And the built-in 10-input mixer helps you come up with the perfect stereo mix.

Designed by engineers, priced by musicians.

The X4 is a great mix of low price and digital performance.

Goodbye, tape hiss, wow and flutter. Hello, 37 minutes of 4-track digital audio.

With the Sony MDM-X4 you're recording onto a digital optical disc. That means improved sonic performance and no cross-talk. No need for noise reduction. No heads to clean. And no generation loss for near CD quality sound.
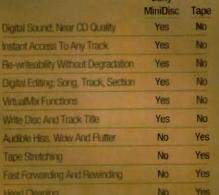
Compare MiniDisc with tape

140MB. 2¹³⁄₁₆ x 2¹⁵⁄₁₆

	Sony MiniDisc	Tape
Digital Sound: Near CD Quality	Yes	No
Instant Access To Any Track	Yes	No
Re-writeability Without Degradation	Yes	No
Digital Editing: Song, Track, Section	Yes	No
VirtualMix Functions	Yes	No
Write Disc And Track Title	Yes	No
Audible Hiss, Wow And Flutter	No	Yes
Tape Stretching	No	Yes
Fast Forwarding And Rewinding	No	Yes
Head Cleaning	No	Yes
Noise Reduction	No	Yes
Crosstalk	No	Yes

and you'll weep at the difference.

Some things actually get better as they wear out. Tape is not one of them.

Tape stretches and deteriorates the more it's used, a fine idea if you're talking jeans, lousy when it comes to tape.

But with the Sony X4, there is no tape. No tape to break. No tape to wear out. No tape to fast forward and rewind. In fact, with the X4 you can rerecord over the same section a million times with no loss of quality, making MiniDisc the cost-efficient way to go. Not to mention the most durable.
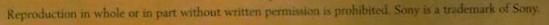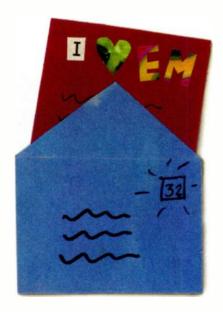
MiniDisc is the way to go digital. Sony is the way to go MiniDisc.

See your Sony dealer for a hands on demo. For more information and the dealer nearest you, call 1-800-635-SONY, ext. X4.

SONY

BUDGET INFO

Thanks for the September **EM**. "The Budget Desktop Studio" gave me a good grounding on the topic, but I have one caveat: the minimum system requirements that you recommended for the Mac are too high. I have been running *Logic Audio* with an original Audiomedia card for 1½ years on a PowerBook Duo 230 with 12 MB of RAM, a Duo Dock, a Seagate Barracuda 1 GB HD, and System 7.5.3. The total purchase cost was $3,100, and I have had only two crashes, with no files lost. I run a system of only 2.5 MB and rebuild the desktop, zap the PRAM, and optimize my HDs regularly.

In addition, I am thinking of buying the new Audiomedia III card, but I was also thinking of buying one of Power Computing's CPUs. So far, no one—including Digidesign—can tell me if this will work. Any ideas?

ambientjaz@aol.com

ambientjaz—Several readers have asked whether they could use a less powerful CPU than I suggested. Of course you can; I never said my Power Mac recommendations were "minimum requirements." (I was careful to use phrases such as "if possible" and "it's wise.") I simply recommended a system that won't break most readers' budgets yet allows for future expansion via PCI, gives high-quality performance, and lets you fully integrate all the recording and processing functions I described. The only "practical minimum" I suggested was the Quadra 650 (or the 840AV) for those who don't want a Power Mac, and even that can be stretched. Crashes aren't the main issue, general performance is. A slower CPU translates into fewer tracks of audio, slower effects processing, and possible playback glitches. Sometimes the performance, although usable, is so sluggish that it makes production more of a chore than a joy.

As to the Audiomedia III, it works as well with the Power Computing clones as it does on comparable Apple Macs. I own a Power Computing PowerWave 604/132 and have used several Apple Power Macs. In my experience, anything that works well with an Apple Mac works well with a comparable Power Computing clone.—Steve O.

TIMELY APPEARANCE

I just wanted to say that "The Budget Desktop Studio" was fabulous and timely. I'm right in the middle of setting up my hard-disk recording environment, and I gained many helpful facts and insights from this article. Thank you for all you do to help us home-studio fanatics. Your efforts are appreciated, and I wish you continued success.

Robin Henry
Midland, Ontario, Canada

THOSE WIN-WIN SITUATIONS

While glancing through "Reviews" in the September 1996 issue, I came across a somewhat bothersome quirk: a couple of products that interface with computers, specifically the Digidesign Audiomedia III card and Yamaha CBX-D5, both have "Win" listed next to their headings to designate their use with Windows-based PCs. My gripe is, in reading the text and/or the specifications summaries, it's clear that these devices also work with PPC Macs. I feel it's a disservice to current or potential Mac users to not modify (or simply delete) this misleading heading at the start of your review.

Dave Kesner
dave_kesner@ucsdlibrary
.ucsd.edu

Dave—As you note, Audiomedia III and the CBX-D5 are available for both Mac and PC. The CBX-D5 also works with the Atari Falcon. The CBX-D5 review's Product Summary (p. 150) includes the system requirements for all three platforms. However, the Mac version of Audiomedia III was not yet shipping when we prepared the review. The text stated that the product "is available for PCI-based Power Macs" because we expected the Mac version to be available by the time we were in print, as indeed it was. But we couldn't be sure about the Mac system requirements, so these specifications do not appear in the review's Product Summary (p. 133).

We initially attempted to name all three supported platforms in the CBX-D5 review title. Unfortunately, the formatting of our reviews section did not permit us to write a long enough headline to do it. Faced with this layout limitation, we felt it was important to at least identify the platform we actually used for our tests—Windows, in the two examples you cite—and mention other compatible platforms in the main text and Product Summary.

This has long been our policy for reviews of cross-platform products. However, if enough readers inform us that our current approach doesn't serve their needs, we will try to find another solution.—Steve O.

PRAISEWORTHY

Thank you for Jennifer Conrad Seidel's piece in the August issue ("Working Musician: Publishing Your Praise"). As a part-time praise and worship leader—and full-time freelance composer-arranger—I'm constantly learning more about the different facets of music production. Keeping up on the latest gear, production techniques, and software would be practically impossible for me without your help.

At my church music MIDI seminars, I keep a stack of **EM** back issues on the info tables and list **EM** on my resource list as *required* reading for the serious MIDI musician. Just want to let you know that the Christian market is a growing segment of your readership. Keep up the great work!

Perry Schjolin
optasia@primenet.com

MICHELE ALANIZ

NO PAYOLA HERE

I was rereading your review of the Yamaha 02R (July 1996) when the pull quote on p. 131 caught my eye: "Once you're in the pristine digital domain, you'll want to stay there." The point the writer makes—that multiple analog-to-digital conversions are undesirable and, with a digital mixer, unnecessary—is a good one. But why the use of "pristine"? I think with that word you do a disservice to your readers. For example, a guy wrote to you in the same issue about mixing to DAT vs. mixing to cassette. He seems to wonder whether the DAT will somehow make his mixes instantly hi-fi, which is pretty understandable when you consider some of the questionable claims made for digital audio over the years.

I recognize that EM is written in a gung-ho style, with more colorful language than you'd find in a purely technical journal, but jaded old-timers like me kind of wonder whether digital audio advertisers pay you a bit more every time you subtly promote the medium with your choice of words. I hope you'll avoid this stealth boosterism in the future; I'll trust your articles more.

**Mitch Easter
dangrman@nr.infi.net**

Mitch—Hey, watch it: I'm an old-timer, too! So, please trust another aging analog baby when I say there is no "stealth boosterism" or clandestine payola from digital manufacturers at EM. Our boosterism is totally upfront; we are "gung ho" about anything that helps our readers create better-sounding music. To that end, we certainly applaud (and support) the digital tools that have made it possible for home recordists to upgrade the sonic quality of their work. But we have not turned our back on analog sound. We've published numerous features on optimizing analog tape, using the new generation of groovy tube toys, and willfully sabotaging that pristine digital sound.

It's no secret that the digital domain—through MDMs, hard-disk recorders, and the new MiniDisc multitracks—has overtaken analog as the medium of choice in home and project studios, and our coverage obviously reflects this situation. Our ma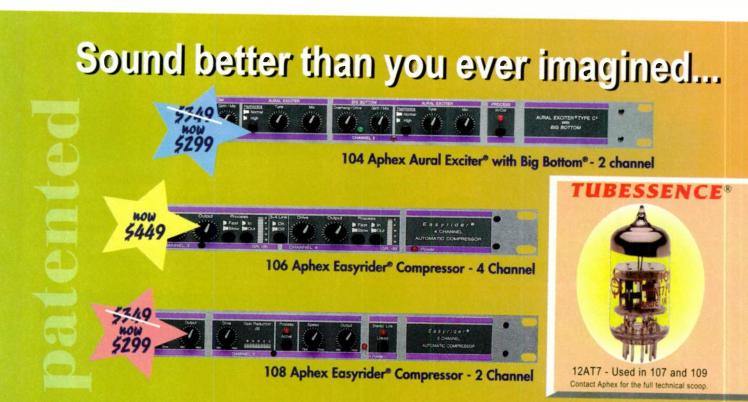in goal, however, has never strayed from informing recordists how to use whatever tools they have at their disposal to produce the sounds they hear in their blissfully twisted little heads. (And, by the way, I absolutely loved the records you produced with your band, Let's Active.) —Michael M.

STAND NEXT TO YOUR FIRE

Thanks for the information on mLAN and FireWire ("Tech Page: Fire in the Wire," July 1996). The last time I checked, http://www.firewire.org didn't have any links to Yamaha or mLAN. Predictably, the interest is mostly in the video realm. Do you know how I could get a copy of the mLAN spec? If enough companies agree on this protocol early enough, this could be a really good thing.

**Jim MacArthur
Lexicon, Inc.
Waltham, MA**

Jim—According to Mike D'Amore at Yamaha, the mLAN spec is available at the FireWire Web site, but it's somewhat buried. A better source is the Yamaha Strategic Business Alliance's Web site (http://www

.ysba.com). This site is open only to qualified hardware and software developers. It takes about 48 hours to complete your registration, after which you have full access to the site. Best of all, membership is free to all qualified applicants (which would undoubtedly include Lexicon).—Scott W.

MIKE IT

I am impressed with the specs of the Crown CM-700 ("Reviews," July 1996). In the review, you gave it a rating of 4 in Audio Quality. Does that mean I could sell my large-diaphragm condenser mic, get two or three CM-700s, and still get the same audio quality? I'm doing more and more projects and I need more mics.

Andy Soriano
andys@mnl.cyb-live.com

Andy—The CM-700 is a wonderful mic, but I wouldn't toss out your large-diaphragm model in favor of three identical small-diaphragm mics. For one thing, a fine large-diaphragm condenser typically delivers a broader, warmer low end than a small-diaphragm condenser of comparable quality.

(That's one reason why premier vocal mics such as the Neumann U87 are large-diaphragm models.) I'd recommend a fair amount of diversity in your mic collection, so think about buying two CM-700s to facilitate stereo miking and holding on to your large-diaphragm condenser.—Michael M.

NO MATTER WHERE YOU ARE

Thank you, **EM**, for the great information in every issue. I am in the U.S. Navy, stationed overseas in Japan, and have been away from the States for three years. Fortunately, **EM** manages to keep me posted on the newest gear and emerging technologies. It doesn't matter whether I am in the middle of the Arabian Gulf, somewhere off the coast of Hong Kong, or in home port Yokosuka, Japan, **EM** is there to inform me.

Christopher Crass
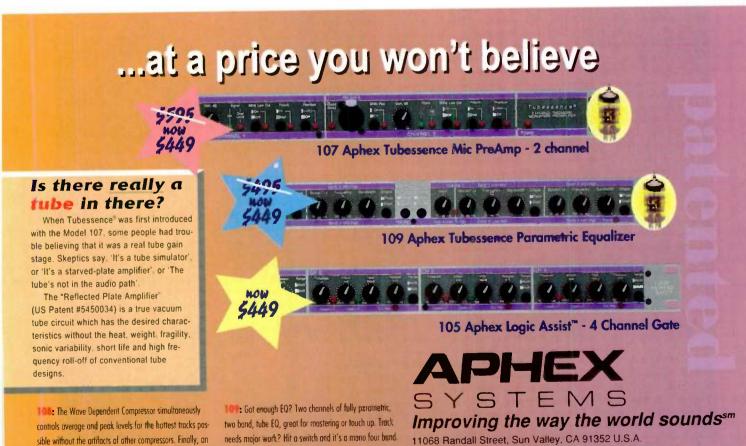pacras14@cv62.navy.mil

SHAKE A TAIL FEATHER

Please explain the science behind "tails out" storage of analog magnetic tape. Either way the tape is wound, there is a layer of plastic backing with the iron-oxide layer toward the center. So why is there less print-through with tails-out storage?

Herb Cronin
Harrisburg, PA

Herb—Tails-out storage does not diminish the amount of print-through. What it does, however, is ensure that any print-through is heard after the original signal as a faint echo. This is preferrable to preprint-through—which can be caused by heads-out storage—where a ghost sound is audible before the source signal. A classic example of this somewhat disconcerting effect is Led Zeppelin's "Whole Lotta Love," where Jimmy Page's introductory guitar riff appears before the song actually kicks in.—Michael M.

David Carson
Graphic Artist
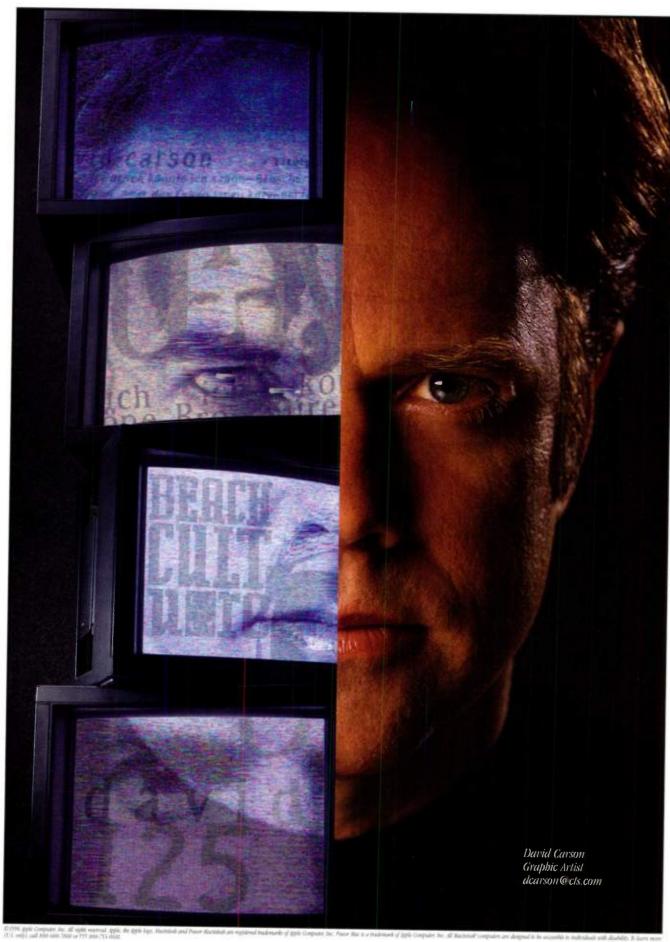dcarson@cts.com

SUDDENLY THERE'S AN EXPLOSION
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With words, with movement, with a heart-stopping new power like never before. *You don't know*

where it came from. Or where it's going. But you know how it will get there.

MACINTOSH.
Master the Media.

Inspiration happens. In David Carson's case, inspiration happens a lot. Fortunately for David, what he sees on his screen is never very far from what he sees in his mind. Power Macintosh computers help him explore just about anything he can imagine. His groundbreaking design work for *Surfer* and *Ray Gun* magazines – as well as for clients like AT&T, Nike and Pepsi – confirms it. But the Power Mac™ is also flexible enough to change when David's ideas do. So he can take his concepts and easily reexpress them in new mediums – from film to the Internet. It's the kind of freedom that allows him to push not only the limits of his computer, but the limits of his creativity. Authoring web sites. Manipulating graphics. Altering video. It's never been easier for David to create the work he's known for: the kind that never existed before.

To find out more about our cross-media authoring solutions, visit us at www.masters.media.apple.com or call 800-305-0374 for information via fax.

WHAT'S NEW

By the EM staff

▲ ALESIS NANOCOMPRESSOR

Although palm-sized 24 × 8 mixing consoles are not yet available, in the meantime you might enjoy getting your hands on the latest smaller-than-a-sandwich piece of audio gear from Alesis, the NanoCompressor ($149). Weighing only 1.25 pounds, this stereo compression unit fits neatly into a one-third rackspace package but can put the squeeze on Goliath-sized signals.

The NanoCompressor's front panel includes controls for ratio, threshold, attack, release, output level, and bypass. A 6-segment LED meter can be switched to register input or output levels while a second meter displays gain reduction. The rear panel includes ¼-inch stereo inputs and outputs as well as a ¼-inch TRS sidechain input.

Stereo channels on the NanoCompressor are internally linked, which means you can't control them separately. However, the unit does offer the choice of RMS or peak compression styles as well as hard-knee or soft-knee compression curves. Alesis Corporation; tel. (310) 841-2272; fax (310) 836-9192; e-mail alecorp@alesis1.usa.com.

Circle #401 on Reader Service Card

▶ SPIRIT FOLIO SX

The concept of portability took on new meaning with Spirit's original Folio mixer, which was outfitted with a carrying handle. The new Folio SX ($769) carries on the tote-handle tradition and offers many new features, as well. (The handle doubles as a tilt for the control surface or can be removed to allow for optional 10U rack-mounting.) The streamlined, aluminum-framed, 4-bus board provides twelve mono mic/line channels and four stereo channels for a total of twenty channel inputs. You get stereo main mix, subgroup, and monitor outs; one mono out; eight channel direct outs; and three aux sends. The direct outs can be switched pre- or postfader.

All twelve mic/line channels feature mic preamps with 60 dB gain range; insert points; switchable, 18 dB/octave low-cut filters set at 100 Hz; 3-band EQ with sweepable mids; three aux controls (one prefader, one postfader, and one switchable pre/postfader); channel on, PFL, and mix/subrouting switches; and 100 mm faders. The stereo channels include 100 mm faders; four balanced ¼-inch inputs, each offering gain and aux controls; four RCA inputs with rotary level controls and mix/sub routing; 2-band EQ; and channel on, PFL, and mix/subrouting switches.

The Folio SX also offers a headphone jack; insert points for the main mix; switchable, global, +48V phantom power; and 2-track returns on RCA jacks. Two 3-color, 10-segment bargraph LED meters can be switched to show mix output, PFL, or AFL solo levels. A latched power connector ensures that the detachable power cord remains secure.

The compact board boasts impressive specs: frequency response is rated at 20 Hz to 30 kHz (±1 dB), channel crosstalk at <95 dB (@1 kHz), and THD <0.006%. Spirit by Soundcraft, Inc.; tel. (916) 888-0488; fax (916) 888-0480; Web http://www.Spirit-by-Soundcraft.co.uk.

Circle #402 on Reader Service Card

▼ PEAVEY DELTAFEX

If patch-surfing through a thousand presets is not your idea of fun and all you really need are the essential digital-quality stereo effects, take a look at Peavey's Deltafex ($159.99). A front-panel knob selects among sixteen effects: chorus, phaser, flanger, seven reverbs, four delays (including stereo ping-pong delay), a parallel reverb and delay, and rotary-speaker emulation with morphing speed control. Each effect has two pa-

rameters controlled by dedicated knobs.

The unit features knobs for input level, output level, and wet/dry mix and has a bicolor clip LED. A jack on the back panel admits a footswitch that serves as a speed controller for the rotary-speaker emulator and functions as an on/off switch for all other effects. I/O is on unbalanced ¼-inch jacks. Peavey Electronics; tel. (601) 483-5365; fax (601) 486-1278; Web http://www.peavey.com.

Circle #403 on Reader Service Card

THE SHORTEST DISTANCE FROM INSPIRATION TO REALITY.

Introducing the new family of Power Macintosh computers.

We know you're busy, so we'll make this fast. Introducing the new family of Power Macintosh computers. More power. More speed. More freedom. More Macintosh. The easiest, most powerful way yet to unleash and communicate your ideas. And, on average, 50% faster than just a year ago.*

In fact, the new Power Macintosh 9500/180MP features the performance of not one, but two RISC-based PowerPC 604e processors running at 180 MHz, providing the muscle needed to execute the computing-intensive tasks you need to get done — from video manipulation to 3-D modeling to sound editing.

We've built it with an internal 8x-speed CD-ROM drive, 2 gigabytes of hard disk space, 6 PCI slots, an upgradable processor as well as all the software you need to get up and running on the Internet. So when your business grows, your Power Mac grows with you.

Bottom line, it all happens faster on a Power Mac. Which means you spend more time exploring your ideas, and less time waiting for your computer to catch up with you. Vision accomplished.

REV UP! ▲ ▲ ▲ ▲

▼ CAKEWALK MUSIC SOFTWARE

Competition is flourishing in the Windows sequencer market, and Cakewalk Music Software has responded by upgrading its entire line to include digital audio recording and 32-bit Windows 95 versions: *Cakewalk Pro Audio* 5.0 Deluxe ($479), *Cakewalk Pro Audio* 5.0 ($399), *Cakewalk Professional* 5.0 ($249), *Cakewalk Home Studio* 5.0 ($129), and *Cakewalk Song Station* 5.0 ($69). Each program comes on a CD-ROM with Windows 95 and Windows 3.1 versions. Upgrade pricing varies depending on what version you have; contact the manufacturer for details.

Cakewalk Pro Audio lets you record as many audio tracks as your hardware

can handle. *Cakewalk Professional* and *Cakewalk Home Studio* can record four audio tracks, and *Song Station* can record two tracks. Audio can be recorded through any Windows sound card, and the *Pro Audio* versions support Digidesign's Audiomedia III card and Soundscape's SSHDR1. All versions except *Song Station* offer a 10-band graphic EQ. The *Professional* and *Pro Audio* programs sync to SMPTE or MTC.

The entire line features a Track view that lets you cut, copy, and paste clips of MIDI or audio data. *Cakewalk Pro Audio* includes parametric EQ and lets you groove-quantize audio to MIDI, MIDI to audio, and audio to audio; extract timing; remove silence; and draw custom fade curves. All versions have 128 levels of undo/redo.

Cakewalk has added several new instrument templates, including Alesis DM5, DigiTech Studio Vocalist, Korg Prophecy, and Novation BassStation definitions. A Virtual Jukebox plays MIDI and WAV files. Soundtrek's *Jammer Hit Session* auto-

accompaniment software is bundled with the program. All versions except *Song Station* provide enhanced notation features, including guitar chord grids, hairpins, pedal markings, expression text, and lyrics.

Cakewalk's CAL macro language is available in the *Professional* and *Pro Audio* versions. The Deluxe edition also includes the *Musician's Toolbox* CD-ROM, a 600 MB menagerie of samples, groove templates, MIDI files, utilities, and twelve more tutorial movies. Cakewalk Music Software; tel. (800) 234-1171 or (617) 926-2480; fax (617) 924-6657; e-mail sales@cakewalk.com; Web http://www.cakewalk.com.
Circle #404 on Reader Service Card

▶

OPCODE

Opcode has released *Overture* 1.2 ($495; academic version $250; upgrade from v. 1.1 at no charge; upgrade from pre-1.1 version $19.95). The new version allows playback using Apple QuickTime Musical Instruments and accepts General MIDI patch names. The software now allows you to copy chords, crescendos, dynamics, text, and so forth by Option-dragging. Other new

▼ ZOOM STUDIO 1204

Zoom's Studio 1204 ($249) is a discrete stereo/dual-mono effects processor with some unusual vocal effects and MIDI control. It has two banks of sixteen effects, each with sixteen variations, for a total of 512 programs. The unit also offers 100 user memory locations.

The effects include reverb, echo, delay, chorus, flanger, pitch shift, and tremolo. Some programs route delay, chorus, flanger, or pitch shift into re-

verb. In addition, there are seven parallel effects with modulation effects on the left channel and time-based effects on the right. Finally, the Studio 1204 provides vocoding, vocal distortion, rotary speaker simulation, and noise reduction. A 2-band EQ can be applied after any effect.

The parameters can be edited via two front-panel knobs, and you can control the input and output levels and wet/dry mix. Presets in memory can be recalled via MIDI Program Change messages,

and programs can be altered in real time via MIDI Control Changes.

The ¼-inch L/R inputs and outputs operate at -10 dBm or +4 dBm. A ¼-inch microphone input on the front panel connects to an internal preamp. The unit samples at 44.1 kHz and uses 18-bit, 128x-oversampling converters. Zoom rates the unit's frequency response at 10 Hz to 20 kHz. Samson Technologies; tel. (516) 364-2244; fax (516) 364-3888; e-mail sales@samsontech.com.
Circle #407 on Reader Service Card

with crescendo and decrescendo, and support for nested tuplets. Opcode Systems; tel. (415) 856-3333; fax (415) 856-0777; e-mail info@opcode.com; Web http://www.opcode.com.

Circle #405 on Reader Service Card

▶ SONIC FOUNDRY

Version 4.0 of Sonic Foundry's *Sound Forge* ($495; upgrade from v. 3.0 $99; upgrade from *Sound Forge XP* $199) includes several new features and amends many older ones. The program now lets you open Video for Windows (AVI) files to edit sounds in sync to video frames, and it can send and receive MTC and SMPTE time code during recording or playback. Other new features include an unlimit-ed Undo/Redo history; support for Java, RealAudio, and OLE; 4-band compressor/limiter; crossfade-loop tool; and peak-level meter. In Direct Edit mode, you can make destructive edits directly to the file you're working on, which re-sults in faster opening and saving but means that you can't revert to the original file.

During recording and playback, *Sound Forge* 4.0 checks to make sure that your computer is keeping up with the audio. If not, you can set the program to drop markers in the file or simply stop. You can also drop markers while recording, and a host of new func-tions are included for manipulating markers and regions. The audio-event locator works like a "scrub" fea-ture: playback follows mouse movement when you drag across the Overview strip and loops while the mouse is still.

New reverb, time compres-sion/expansion, pitch shift, phaser, flanger, and wah-wah algorithms have been added. You also get new 4-band para-metric EQ and high- and low-pass filter algorithms. If you

tweak a parameter while listening to a preview, the program automatically generates and plays a new preview.

Sound Forge's interface has also evolved. Vertical and horizontal rulers display level and time values, and new Zoom functions have been added. You can save and load workspaces, and the program can be set to automatically open to the files you were working on when the program was closed. *Sound Forge* 4.0 is available for Windows 95, Windows 3.1, and Windows NT. Sonic Foundry; tel. (800) 57-SONIC or (608) 256-3133; fax (608) 256-7300; e-mail sales@sfoundry.com; Web http://www .sfoundry.com.

Circle #406 on Reader Service Card

Overture's score window showing tear off palettes.

▶ ZEFIRO ACOUSTICS ZA2

Has your DAT deck been properly in-troduced to your Windows comput-er? The ZA2 audio card ($495) from Zefiro Acoustics is designed to estab-lish a working relationship between the two. This ISA-bus card has S/PDIF I/O via RCA or Toslink connectors as well as AES/EBU I/O and stereo RCA analog outs for consumer-level monitoring. The card provides direct digital recording at 32, 44.056, 44.1, and 48 kHz. Thanks to its 24-bit DSP chip, the ZA2 allows real-time sample rate conversion (e.g., from a 48 kHz DAT to a 44.1 kHz file). Real-time MPEG audio decoding should be sup-ported by the time you read this, and other applications are in the works.

D/A conversions for the analog outs are 16-bit. The ZA2's THD is rated at 0.01% (typical), its dynamic range at 85 dB (typical), and its frequency response at 20 Hz to 21 kHz when sam-pling at 44.1 kHz.
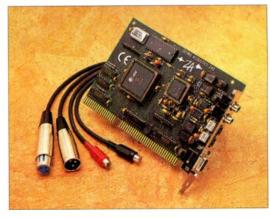
The ZA2 comes with some useful software utilities for DAT recording. One of these allows you to edit CD and DAT subcodes, including SCMS and Start IDs. Another lets you back up 1.2 GB from your hard drive to audio DAT.

The card comes with drivers for Windows 3.1 and Windows 95, and drivers for Windows NT should be available by the time

you read this. Zefiro Acoustics; tel. (714) 551-5833; e-mail info@zefiro.com; Web http://www.zefiro.com.

Circle #408 on Reader Service Card

Now You Can See It.

Mixing Some of our competitors' disk recorders use a portion of their recording LSI to provide mix capability. While this saves money, it can also produce audio artifacts like "zipper" noise when adjusting such critical functions like EQ, pan, and fader level. On top of that, many disk recorders won't even let you make real-time adjustments during mix down, eliminating a critical part of the creative recording process. The heart of the DR mixer is a 16-channel, 24 bit custom LSI designed to provide real-time dynamic digital mix capability. Built-in 99 scene snap-shot automation for all functions and dynamic automation via external MIDI sequencers, combined with 8 or 16 channel 3-band parametric EQ option, ensures that the only limit in the DR Series mixer is your imagination. With its built-in 16 channel mixer, the DR8 becomes the perfect compliment to any 8-track recorder you might currently own. It can mix down its 8 tracks of internal digital audio with an additional 8 inputs from a sampler, tape machine, or a live performance, all in the digital domain. The MT8 mix controller provides a 16 track console format for dynamic remote control of all mix and EQ parameters.

(Monitor/Keyboard/Iomega Drive and Batteries not included.)

SuperView™ We sort of went into a frenzy packing new features into our DR8 and DR16. When we stepped back to take a look at what we'd done, we realized we crammed a whole roomful of equipment into a single 5U box. In order to help keep track of everything that's going on inside our "studio in a box", we developed the SuperView™ SVGA monitor board. SuperView™ mounts internally in the DR8 or DR16 and provides envelope and track information for up to 16 tracks of audio, as well as region highlighting for record, playback, and edit. SuperView™ is further enhanced by 16 track level meters with indicators for left/right master out and aux 1/2 out. The time indicator will read in the same format as the DR front panel. SuperView™ requires no external computer, simply plug your SVGA compatible monitor into a SuperView™ equipped DR Series recorder and you're ready to go. SuperView™ enables real-time video representation of audio status; no waiting for screen re-draws. What you hear is what you see.

ELECTRONIC MUSICIAN 1996 EDITORS' CHOICE

AKAI DIGITAL

Akai Digital
1316 E. Lancaster Ave.
Fort Worth, TX 76102, U.S.A.
Ph. 817-336-5114
Fax 817-870-1271

Keyboard Interface To increase the power of SuperView™ even further, we added an ASCII keyboard input to the SuperView™ card, allowing a standard ASCII keyboard to operate as a control interface for SuperView™ equipped DR Series recorders. Function keys will provide the ability to zoom in on a single track, as well as zoom in/out timewise for precise edit capability. All tracks and locate points can be named, allowing you to manipulate and track large amounts of data in a very simple manner. A unique interface has been developed to allow track arming, transport control, and edit functions directly from the keyboard, providing enhanced productivity through an intuitive human interface design.

SEE US AT AES BOOTH #1019

▶ INTELLIGENT DEVICES IQ

Spending another afternoon hunched over your equalizer trying to match that replacement guitar track to the mix you thought you had finished a month ago? Maybe you should let your Mac take a crack at it. Intelligent Devices' *IQ* TDM plug-in ($749) is designed to alter the spectrum of one take, track, or song (mono or stereo) to match that of another take, track, or song.

IQ uses reference audio (which can be one of your own tracks, a CD track, or an entire CD) to determine the desired spectrum, or it can use templates from a user library. Then it analyzes the signal to be corrected and creates an FIR filter model that runs as a Digidesign TDM plug-in, so it can be inserted anywhere in the Pro Tools audio path. The filter curve can be edited, compressed, or expanded. In manual mode, *IQ* lets you hand draw any filter curve, with as many peaks or notches as you wish.

According to Intelligent Devices, *IQ* has no typical sonic signature and will not introduce phase shift, ringing, or other audio undesirables. Intelligent Devices; tel. (410) 744-3044; fax (410) 788-6370; e-mail intdev@clark.net; Web http://www.richarde.com.

Circle #409 on Reader Service Card

▼ YAMAHA VL70-M

If you've been holding your breath waiting for Yamaha's physical-modeling synthesis to come down to a price you can afford, grab a MIDI controller and get ready to let loose. The new VL70-m ($799.95) is a half-rackspace tone generator based on Yamaha's new VL Version 2 firmware. This monophonic unit features 256 preset programs, 64 user programs (which are edited versions of the presets), and six custom programs created with Yamaha's editing software. Mac and Windows editors can be downloaded from the Internet. An onboard effects processor provides reverb, chorus, and distortion.

The front panel has a stereo headphone jack, an input for a Yamaha WX7 or WX11 wind controller, and an input for a Yamaha BC1, BC2, or BC3 breath controller. The back panel holds ¼-inch L/R outs; MIDI In, Out, and Thru ports; and a serial computer interface. (Windows users can download the required driver from Yamaha's Japanese Web site.) The VL70-m also features a VL-XG mode, which allows you to assign the unit to a Part in an XG setup.

Also new from Yamaha is the half-rackspace P50-m sound module ($499), which features piano sounds from the company's P-series digital pianos. The unit's 6 MB of compressed waveform ROM provide 28 piano sounds, from classic grands to vintage electrics. The P50-m is 32-note polyphonic, but polyphony is reduced to sixteen notes for the eight 2-layer sounds. Multiple units can be connected to increase the polyphony.

The unit includes a 3-band graphic equalizer with 12 dB boost/cut. Using the data knob and buttons, you can set brightness and reverb depth, adjust the master tuning, transpose by ±12 semitones, and select from eight Velocity-response curves. There are eleven reverbs and eleven chorus effects with several editable parameters, which vary depending on the effect. These effects can only be accessed via MIDI and cannot be saved to memory, although the unit retains the current settings when switched off.

In addition to MIDI In and Out jacks, the P50-m has ¼-inch L/R outs and an ⅛-inch, stereo headphone jack. Like the VL70-m, the P50-m can be used in an XG setup. In this mode, it responds to messages for the piano part and passes the rest to the MIDI Out jack. Yamaha Corporation; tel. (714) 522-9011; fax (714) 739-2680; e-mail info@yamaha.com; Web http://www.yamaha.com.

Circle #410 on Reader Service Card

▼ SYMETRIX 551E

The Symetrix 551E 5-band parametric EQ ($449) features five fully overlapping bands, each with dedicated knobs for adjusting frequency (from 10 Hz to 20 kHz), bandwidth, and boost/cut (+12 to -20 dB). You also get a high-cut filter (3 kHz to 65 kHz) and a low-cut filter (6 Hz to 260 Hz) with 12 dB attenuation. Inputs and outputs are provided on XLR and balanced ¼-inch connectors.

Other features include servo-balanced outputs, direct-coupled input/output stages, and an internal power supply. THD+N is rated at <0.002%, dynamic range at >114 dB, and frequency response at 20 Hz to 62 kHz (+0 dB, -3 dB). Symetrix; tel. (206) 787-3222; fax (206) 787-3211; e-mail symetrix@symetrixaudio.com; Web http://www.symetrixaudio.com.

Circle #411 on Reader Service Card

▶ ANTEX STUDIOCARD

In general, your sonic signals will be cleaner if you keep them in the digital domain as much as possible. Antex's StudioCard ($1,595) is a 32-bit PCI card for Windows PCs that provides four discrete channels of digital audio. AES/EBU and S/PDIF I/O are provided via a 15-pin D-sub connector. An optional breakout cable for the 15-pin connector ($50) gives you two XLR jacks (for AES/EBU and S/PDIF), MIDI In and Out ports, linear time code (LTC) in and out jacks, and an external clock input.

A 25-pin D connector provides access to four independent, balanced, analog ins and outs, which operate at +4 dBu or -10 dBV. A special cable (included) breaks the 25-pin connector out to eight XLR connectors. The A/D converters are 16-bit, 64x oversampling Delta Sigma, and D/A converters are 18-bit, 128x oversampling Delta Sigma. The card supports sampling rates between 6.25 kHz and 50 kHz (in 50 Hz increments). Its onboard DSP allows digital mixing in real time.

The StudioCard offers an array of synchronization options. It receives and generates 24, 25, 29.97, 30 fps nondrop, and 30 drop-frame SMPTE time code. It also can sync to MTC (via an MPU-401 compatible MIDI port), the host clock, or external word clock.

You can install as many as four StudioCards in one PC, giving you up to sixteen channels. In addition, the card has an expansion connector for optional daughtercards (not yet available). The dynamic range is rated at ≥92 dB. Antex claims a frequency response of 20 Hz to 20 kHz (± 0.1 dB) and THD of <0.003%.
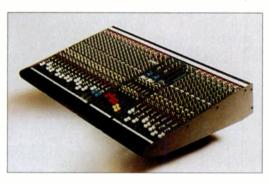
The Plug 'n' Play-compatible Studio-Card comes with drivers for Windows 95 and Windows NT. It requires a Pentium PC with a Plug 'n' Play BIOS, PCI bus, and 16 MB of RAM. Antex; tel. (800) 432-8592 or (310) 532-3092; fax (310) 532-8509; e-mail info@antex.com; Web http://www. antex.com.

Circle #412 on Reader Service Card

▼ ALLEN & HEATH GL2000

When it comes to electronic gear, it's not uncommon to see prices going down even while quality improves. Such is the happy fate of Allen & Heath's GL2000 sound-reinforcement console, an updated version of the company's GL3. The GL2000 is a 4-bus board available in three sizes: the 12-channel model 412 ($2,495); 16-channel model 416 ($2,995); and 24-channel model 424 ($3,995). The GL2000-424 retails for nearly $1,000 less than the 24-channel GL3.

As with the GL3, each GL2000 channel has mic and line inputs, mic preamps, an insert point, a low-cut filter, a phase-reverse switch, and a phantom-power switch. The bypassable, 4-band channel EQ has sweepable high-mid and mid bands. You also get mute and PFL switches and 100 mm faders on each input channel. The six aux sends are switchable post- and pre-EQ/fader in groups of two and four. The console features balanced XLR main and monitor-mix outputs. A dual-function capability allows the console to be configured either as a front-of-house sound-reinforcement mixer or as a dedicated monitor board. The input channels and master channel are modular so they can be individually serviced.

Unlike the GL3, the GL2000 offers two stereo inputs, 4-segment LED meters (located next to the faders) on each channel, rear-panel (rather than top-mounted) input connectors, and an internal power supply. Two of the six stereo returns offer 2-band EQ, routing switches, panning, and faders. The master section includes 2-track send and return, 12-segment LED meters, and individual pan controls for each subgroup. Two or more GL2000s (as well as the GL2, GL3000, and GL4) can be linked together at the bus level with optional Sys-Link cables.

The console's frequency response is rated at 20 Hz to 50 kHz (+0/-1 dB) and adjacent-channel crosstalk is <94 dB (@1 kHz). THD+N (at unity gain) is 0.008%. Allen & Heath U.S.; tel. (801) 568-7660; fax (801) 568-7662.

Circle #413 on Reader Service Card

▶ APOGEE ELECTRONICS FC-8

As every EM reader knows, the two leading modular digital multitrack tape recorders are TASCAM's DA-88 and Alesis' ADAT. These two systems use mutually incompatible tape and digital-audio interface formats, so you need a converter to move eight tracks of digital audio from one type of MDM to the other.

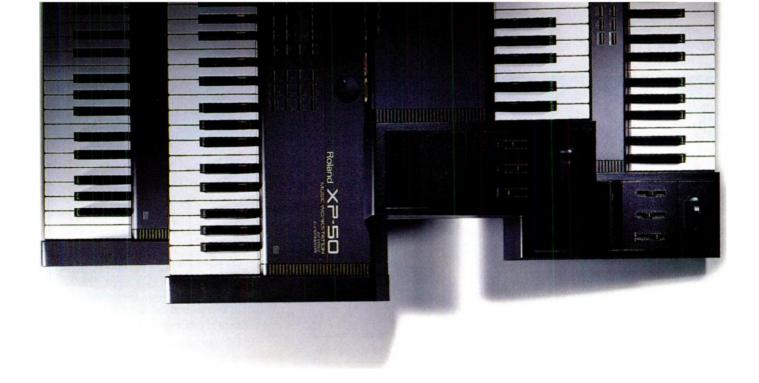Apogee Electronics' FC-8 format converter accomplishes this task at a bargain price ($549). The FC-8 has a DB25 connector for the bidirectional TDIF (TASCAM) interface and an input/output pair of optical ADAT connectors. A BNC connector lets you send word clock to the DA-88 when it is in Slave mode.

The converter is fully bidirectional, and either MDM deck can be the master. The 5 × 2-inch box is powered by an external supply. Apogee Electronics; tel. (310) 915-1000; fax (310) 391-6262; e-mail converters @apogeedigital.com; Web http://www .apogeedigital.com.

Reader Service Card not available

There are a lot of powerful workstations in the world.
Most of them are XP-50s.

In the short time since we introduced the XP-50, it's gone on to set quite a few standards. Like innovative performance features, value, and sheer expressiveness to name a few. And then, of course, there's popularity. Last year the XP-50 was the top choice of musicians around the world. Makes sense, though, because from Real Time Phrase Sequencing (RPS) to 640 onboard patches to its powerful sequencer, this workstation has it all. Plus, it comes with 4 expansion board slots, the most in the industry. And to fill them, our ever-expanding library now has 9 unique expansion boards for the ultimate in expandability. So stop by your nearest Roland dealer today and see the new standard in workstations.

⌐Roland®
XP-50 Music Workstation

SEE US AT AES BOOTH #701

WRH

SOUND ADVICE ▲ ▲ ▲ ▲

▶ FRONT ROOM PRODUCTIONS
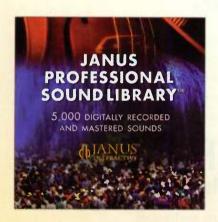
Front Room Productions' *Producer Series* vol. 1 CD-ROM ($199.95) delivers more than 500 MB of vintage synthesizer samples for the Kurzweil K2000 and K2500. Sampled synthesizers include the PPG Wave 2.3; Sequential Prophet VS; Roland MKS-80, MKS-30, and MKS-20; and Yamaha CS-80, DX7, and TX7. Analog bass, brass, strings, pads, clavs, comps (sounds with a short, quick attack used to "comp" chords behind a soloist), and electric piano sounds are featured. Front Room Productions; tel. (718) 449-2129; e-mail thefront@interport.net; Web http://www.interport.net/~thefront/index.html.
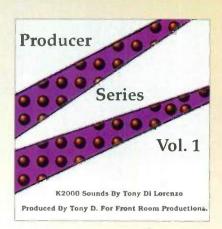
Circle #414 on Reader Service Card

Producer Series Vol. 1

K2000 Sounds By Tony Di Lorenzo

Produced By Tony D. For Front Room Productions.

▲ JANUS INTERACTIVE

The *Professional Sound Library* ($699) from Janus Interactive is a 6-disc CD-ROM set featuring 5,000 special effects, ranging from nature sounds to cocktail-party chatter to science-fiction effects. There are more than 40 categories, such as "Dog," "Household," and "Airplanes," each containing dozens of individual sounds. The sounds were recorded directly to 16-bit, 48 kHz DAT and then downsampled to 44.1 kHz and formatted as AIFF files. All of the sounds are looped.

In addition, the set comes with proprietary editor/librarian software for auditioning and editing the included sounds. The program has a playlist, a search engine that adds its finds to the playlist, transport controls, and a graphic editing window that lets you stretch, cut, paste, and reverse sounds. Files can be converted to WAV or Macintosh SND format, from 16-bit to 8-bit, and from 44.1 to 22 kHz.

The software runs under Windows 3.1, Windows 95, or Macintosh System 7. Janus recommends an 80486 or better PC or a Centris 610 or better Macintosh. Both the PC and Mac versions require 8 MB of RAM. Janus Interactive; tel. (800) 766-0835; fax (503) 690-6690; e-mail oca@janusinteractive.com; Web http://www.janusinteractive.com.

Circle #415 on Reader Service Card

▶ RAREFACTION

Since OSC sold its assets last December, you may have been wondering where to turn next time you need a sound like "Fried Tooth Fairy" or any of the other aural abominations flaunted by OSC's *Poke in the Ear with a Sharp Stick* collections. If so, you're one sick duck, but we know some people who can help. Rarefaction, a new company started by former OSC employees Ron MacLeod and Paul Korntheuer, is now handling the *Poke* series as well as some releases of their own.

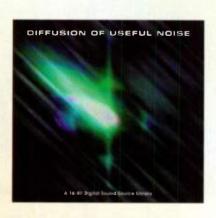New from Rarefaction is *Diffusion of Useful Noise* ($149) by sound designer Keith Hillebrandt, one of the architects of the *Poke* series. This CD-ROM, in 16-bit, 44.1 kHz AIFF format, features over 800 samples broken up into eight categories: Alternate Atmospheres, Ambient Textures, Construction Kits, Fx Machines, Melodics, Nether Vox, Percussivities, and Rhythmic Loops.

The collection includes some found sounds, and others were generated by various synths and software. But don't expect to recognize any.

The Construction Kits include fifteen groups of three or four samples spliced together to form small compositions. Fx Machines are heavily processed sounds, rather than "realistic" sound effects. The Melodics are pitched sounds for assembling songs, although some are more playable than others. Nether Vox and Percussivities feature tortured voices and drums (and other things being hit). The Rhythmic Loops are made up of various sounds processed to give them the rhythmic qualities of drum loops with tempos from 87 to 140 bpm.

Performance and "sustain" loops are trimmed end to end; there are no countoffs or trim-marker beats. All samples are individually normalized with zeroed end points, and according to Rarefaction, all noted tempos are sample-accurate. Downsampled demos are available on the company's Web site. Rarefaction; tel. (415) 346-1840; fax (415) 346-0171; e-mail paul@rarefaction.com; Web http://www.rarefaction.com. ⊕

Circle #416 on Reader Service Card

JANUS PROFESSIONAL SOUND LIBRARY™

5,000 DIGITALLY RECORDED AND MASTERED SOUNDS

JANUS INTERACTIVE

DIFFUSION OF USEFUL NOISE

A 16 Bit Digital Sound Source Library

WRH

PRO FILE

Music Man

Dan Shimmyo makes a solo **Metamorphosis.**

By Diane Lowery

Boy, is Dan Shimmyo a great guitar player. He sizzles through his own compositions and buzzes through Johann Sebastian Bach's Invention no. 8 and Invention no. 13 in ways the composer never imagined. But Shimmyo is also a great drummer, a great bass player, and a great keyboardist. So for his debut album, *Metamorphosis*, Shimmyo simply turned himself into a great band and played all the instruments himself.

To accomplish this feat, Shimmyo made heavy use of Macromedia's *Deck II* 2.5 audio recording software. For example, on Invention no. 13, a piece Bach composed for piano, Shimmyo found that some of the notes were too low for the guitar to play in its standard tuning. Luckily, the hard-disk recorder allowed him to detune his guitar, play back the problematic sections, and seamlessly punch in the lower notes.

"I was able make these punches sound really smooth by programming very short crossfade times in *Deck*," explains Shimmyo. "Now, even *I* can't tell where I punched in. And consid-ering how difficult the parts were, the process was relatively easy. I would learn a few bars, practice the section ten times or more, and then record it."

When working as his "band's" drummer, Shimmyo used a Boss DR-660 drum machine, sequencing the grooves with Opcode's *Studio Vision AV.* "Because *Studio Vision* displays all the notes and note durations, I can really get in there and fine-tune the performance. For example, if a drum fill sounds too machine-like, I'll make some beats softer and more dynamic to emulate the technique of a human drummer. I'll even move some notes *off* the beat, because a lot of drummers get excited when they play fills and consequently throw the rhythm a little out of time."

In another case of time "management," keyboardist Shimmyo added excitement to some static, monaural synth parts by recording them into *Deck* and processing the tracks to create a stereo image.

"I'd copy the part onto another track, so I'd have two identical tracks that could be panned hard left and hard right," he explains. "But then I'd click on the clone track and move it around until it was approximately 20 to 50 milliseconds behind the original track. The slight delay creates an animated stereo image that seems to float around the mix, instead of just sitting between the speakers. The effect sounds really cool."

Although *Metamorphosis* evolved from the head and hands of Shimmyo alone, the record sounds as tough and passionate as a full-band project. It's obvious that, in the right hands, desktop audio tools can make it easier for solo recording artists—and musical schizophrenics—to cover all of the bases. Just imagine a world where you never have to argue with your bandmates about chipping in for rehearsal-studio rental. Now, *that's* beneficial technology.

For more information, contact Cutting Edge Records, PO Box 667, Amityville, NY 11701; tel. (516) 795-0238; e-mail cerecords@aol.com; Web http://www.panix.com/~dshimmyo/edge.html.

Dan Shimmyo

Shop Talk

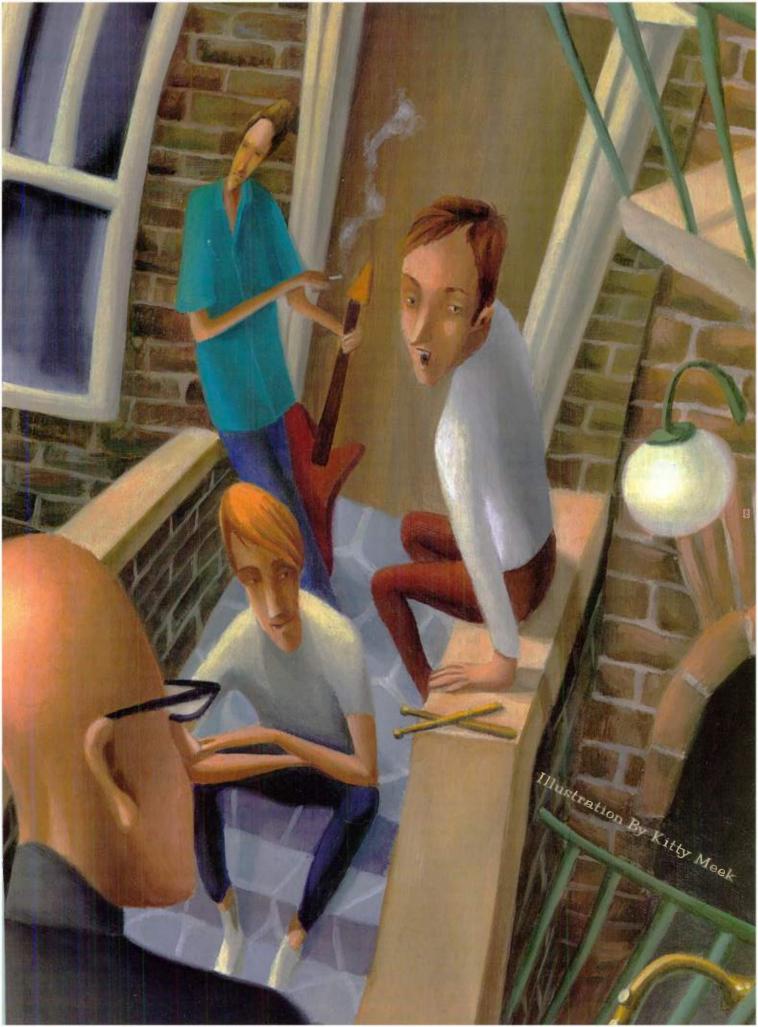**Three noted producers have a heart-to-heart about the psychology of making records.**

There will be no justice in this world until the unsung studio technician who left a 2-track running while the Troggs recorded their follow-up to "Wild Thing" is inducted into the Rock and Roll Hall of Fame. For one thing, the infamous recording of prattle between the Trogg's warring band members is said to have inspired what is possibly the most brilliant rock film of all time, *This Is Spinal Tap*. But more importantly, the Troggs tape establishes for then, now, and all time why God made record producers.

Skeptical? Well, in one part of the Troggs tragi-comedy, the band breaks for a meeting in the control room after fumbling through a series of hysterically inept attempts at recording drum and guitar tracks. One of the wiser members suggests that they hire a producer to help with the sessions, arguing that they need someone who can come in and "sprinkle some fairy dust." In response, a band mate growls, "Well if it's fairy dust you want, I'll [urinate] on the [bleeding] tape!" Needless to say, the Troggs didn't burn up the charts with their next record.

Today, if I meet a young band considering recording their first record without a producer, all I have to do is play the Troggs tape. First they laugh. Then they ask for a list of local producers. It never fails. Unfortunately, many musicians still seem to view record producers as deranged dictators or clueless bumpkins. Those cats *do* exist, of course, and some of them have even been responsible for the biggest hits ever committed to tape. Go figure.

By Michael Molenda

Illustration By Kitty Meek

Shop Talk

However, the vast majority of producers are level-headed artists who *can* sprinkle some sorely needed "fairy dust" on a recording project. To shed some light on the art of production, I interviewed three diverse talents: the legendary Shel Talmy, alternative-music hero Youth, and rising remix star Seibold. I suggest that you steal a few riffs from these learned professionals, lest *your* next tape give the Troggs a run for their money.

Okay, it's a total cliché to say that Seibold follows the beat of a different drummer—and, yes, it's also an annoyingly cute way to describe a dance remix producer—but it's *true*. As frontperson of the aggressive, confrontational Hate Dept., Seibold has shaken up meek audiences and broken his share of dance-music rules. He follows his own muse in a field where genres are defined by extremely specific beats and grooves, and he still manages to be in demand as a remixer. His stylistic isolationism has certainly not stopped his "obnoxious" dance tracks from thumping in clubs everywhere.

In addition, Hate Dept.'s debut album bullied its way out of the underground dance scene to capture a spot on *Rolling Stone*'s Top Ten Alternative Records list for 1995. That's heady stuff for an independent act on a small label. No doubt about it, Seibold has the chops to do his own thing and make it work.
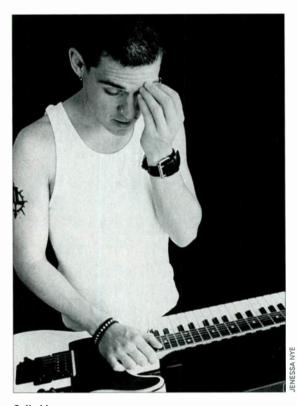
Where do most of your gigs come from?

Well, typically an independent record label will hire me—usually because of my dance-club credibility with Hate Dept.—to make a dance track out of a song that wasn't originally a dance track. That is a big responsibility because if the original song was done by people who had no intention of creating a dance track, then they weren't even thinking about keeping something solid on the floor for people to dance to. It's my job to turn that song into something that makes people dance, and that's exactly what I have to deliver.

I assume that, as with most remix producers, the only thing you keep from the original song is the vocal?

I pretty much rebuild the song, but I'll often use more than just the lead vocal

Seibold

in the remix. I'll have the label send me a DAT with the vocals, a guitar track—if it's a guitar-based song—and some other sounds that gave the original song its character.

As a remixer, I'm very respectful to the original artist. I always try to deliver the goods that the artist originally intended, but I do rhythmic things that keep the song solid for the dance floor. I'm a songwriter myself, and I wouldn't feel comfortable *not* conveying someone else's message.

In most cases, I even maintain the original structure of the vocal—the verse, prechorus, chorus, and bridge sections—intact for the remix. I sample the vocals from the DAT and fly them into the new sequence that I've created for the dance mix. All the elements from the original recording are wild-synched back to my sequence because it's more hassle than it's worth to have the computer chase time code.

I've talked to remixers who claim they must use certain beats, tempos, and sounds to get airplay in, say, an ambient club. Is having to deal with such exacting formats a drag for the remix artist?

If someone tries to tell me there's a formula to what makes people dance—well, that's a hideous joke. People just

SEIBOLD'S MIX MANIA

How can a remix artist get an accurate sense of what his or her mixes will sound like in a dance club stocked with subwoofers and massive horn arrays when the typical project studio is equipped with small, close-field monitors?

"Oh, yeah. It's total weirdness," admits Seibold. "I have my own methods for trying to simulate club sound systems, but I don't know that I've found a good one yet. For example, I'll try to get some point of reference to what it sounds like standing outside of a nightclub by turning up my TOA 380ME monitors as loud as I can and then walking outside my studio and shutting the door. Sometimes I even stand in my closet to listen to mixes. But, ultimately, sitting in front of studio monitors is nothing like standing in a dance club: the club doesn't sound anywhere near as good. I just try to tailor my mixes to what I know will be bad sound rather than struggle to achieve a nearly perfect audio spectrum that the club systems can't reproduce anyway."

dance. Look at the huge resurgence of 1970s disco; it's the comeback of groove music. Now, technically, the production values of that stuff are nowhere near the level of modern dance music. The '70s tracks don't have the sublows, the honking mids, or the pristine highs, but people still get out and get down to it.

Dance music is purely a beat and purely an energy, and all the little production tricks that remixers tend to get tied up with are rather academic. I'm impressed by those production values, sure, but they don't make me want to dance any more or any less.

In addition, I don't get caught up in all the expressions used to describe the different types of dance music. There's *techno, acid house, acid break, jungle,* and all these different words flying around. Now there are subgenres of all the original genres, and each genre is defined by specific grooves and tempos. I'm simply intimidated by all these labels. And to be honest, if I walked into a dance club, I couldn't tell the difference between those genres even if you warned me about what the differences were going to be. To me, all that stuff is just good dance music.

How do you typically construct a remix?

I start compiling sounds first. Electronic musicians always get criticized for using factory patches, and that has always scared me because, early in my career, I didn't know any better, and I used factory sounds to write songs all the time. Now, I don't even like the idea of people recognizing my gear. I create a number of custom patches— typically starting with bass sounds— that are not necessarily intended for a specific track. I do it just to start the process and inspire some ideas.

For the critical kick-drum sounds, I don't really use the classic techno stuff such as the Roland TR-909. I look for big, obnoxious kicks, and the source for those drums is almost always my Roland R-8. I'll distort the sound a little bit, do some radical EQ tweaks, run the signal through a guitar amp, sample it, and that will be my kick drum. I guess I love that drum machine so much because it always sounded bad for my speakers; I could really shred a tweeter if I turned it up too loud.

As far as the rhythm sequences go, I'm not as much of a four-on-the-floor kind of remixer as I used to be. I'm getting much more into programming my own beats rather than using sample CDs with James Brown drum loops or something. There are so many break-beat CDs out there that I'd be afraid of walking into a club one night and hearing a track that uses the same loop that I used for one of my remixes. If I *do* use something from a sample CD, I'll use it in an offensive way. For example, I'll take the first three beats of the pattern and then slow down the fourth beat so it gets all wrong sounding. Basically, I'll wreck the pattern.

When you are reconstructing a song, do you keep any creative ideas from the original production?

I really do try and stick to the original feel for my remixes. If I'm remixing a song where the chorus gets big and raucous, I very much go for that same thing on the dance floor. During the chorus, I'll have big analog synth sweeps going, and then I'll add some extra hi-hats to make everything seem extremely bright and kind of swirly. If the original song goes the opposite way and the chorus becomes quieter and

more introspective, I'll have all my big drums drop away to just an old Roland TR-808 drum sound. The one thing I keep constant is some heavy rhythm that isn't confusing on the dance floor.

Actually, maintaining the dynamic feel of the original song is really difficult. The playback is so loud inside a dance club that, after a while, everything above a certain volume level starts sounding the same. Even when I hear songs that I'm familiar with, I often have a hard time picking out the definition of the chords and the overall tones because it's just so loud. To combat this, I tend to create some dynamic interest with timbre and gain. For example, I'll use thinner, smaller drum sounds for a section where I want the dynamics to "come down." This is really effective because when the heavier beats come back in, it feels like you've been kicked in the butt.

Do you ever grill the club deejays for ideas about what the dance crowd wants to hear?

Shel Talmy

I have friends who are deejays, and they always give me the best and the worst advice. They'll say something like, "Give me sixteen bars of lead-in with just drums and sixteen bars of lead-out so I can segue into another track." I mean, that's good advice, but I'm still left with whatever happens in the middle, and what they've done is take the song value out of the work. How are you supposed to begin and end with sixteen bars of just a beat and still have a song develop in the middle? That's kind of awkward from a songwriting sense. It seems that I always end up trusting my own instincts.

Shel Talmy's production credits are all over classic singles by The Who, The Kinks, Easybeats, Chad and Jeremy, Manfred Mann, Small Faces, and countless other acts from the 1960s British invasion. Back then, Talmy was identified so strongly with the sound of English rock that I assumed he was a Brit himself. Not so. The famous expatriate started out engineering surf-music sessions in his native Chicago until wanderlust brought him to England and a career as one of that country's first independent record producers.

Talmy recently ended a long hiatus to produce the debut album for Nancy Boy (*Nancy Boy,* Sire Records), a much-ballyhooed glam act that sounds retro enough to slip effortlessly into the producer's resume. To complete the '60s connection, the band includes vocalist Donovan Leitch, the heir of flower-power troubadour Donovan, as well as guitarist Jason Nesmith, the son of—you guessed it—Mike "Wool Hat" Nesmith of the Monkees. Happily, the success of *Nancy Boy* has brought

Talmy's wonderful production sensibilities into the here and now, and he has returned to the studio full-time.

How would you describe the Shel Talmy method of record production?

I'm a hands-on producer. I start each project from scratch and stick around until the record is done. The artist and I review the material together, I help choose the songs we'll record, I assist with the musical arrangements, and I go to all the preproduction rehearsals. When I walk into the studio, I'm about 90 percent sure of what I'm going to come out with at the end. In fact, I already have a final mix in mind. I try to leave 10 percent unplanned to accommodate the hopefully wonderful things that happen spontaneously in the studio.

Is there a basic formula to producing a record, or does each project require different methods?

Record production requires a lot of planning and check-and-balance types of decisions to determine what goes on which track and how all these various tracks are going to fit together when you finally get to the mixing stage. Beyond that rather vague explanation, I don't think there's any way to walk someone through the process. I know there are a ton of [production] classes out there now, but I don't know how you teach someone to produce a record.

Having said that, I did learn audio engineering and started out as an engineer, but I gave that up in the 1960s to concentrate on production. It's just too hard to engineer and produce simultaneously because engineers and producers listen to the music in different ways.

That's an interesting point—especially because the personal-studio revolution, by definition, impels many artists to engineer and produce their own work.

Well, I believe it's almost impossible for most people to do both. You can never maintain any objectivity. You need an outside opinion. If I were a serious artist with a home studio, I'd try to get someone in to do the engineering so I could concentrate solely on the creative part.

On that topic, home and project studios allow recordists almost unlimited time to

tweak their productions toward hopeful perfection. Is this a good thing or a bad thing?

It is a mixed blessing. I don't love spending tons and tons of time tweaking something to death. I'm not good at wringing the emotion out of a piece of music until there's nothing left, and I'm not good at building up a record one instrument at a time, because I just don't see the point in it. I mean, we're talking about music here, not mathematics.

I know producers who are as anal as you can possibly get. They're happy to sit for days and listen to every stroke on the hi-hat. Give me a break! These people take months to do a record. I couldn't do that—I'd die of boredom for openers. The Nancy Boy record was recorded and mixed in twenty working days, and we weren't rushing anything.

They were simply very good musicians, and they had the same attitude as some of the '60s bands: let's get in there, get it right, get it done, and not spend money that is going to be taken out of our royalties.

We didn't pull all-nighters to make that 20-day schedule, either. I've never believed that working around the clock is very productive. We worked eight or ten hours a day, went home, and then started again the next day. By keeping reasonable hours, we saved lots of time and effort, and we got better tracks because no one was overtired. It's simple, really: when the energy level starts fading, it's time to call the session and get some sleep.

Of course, to get these tracks so easily, it helped that Nancy Boy is a bit of a throwback—as are a lot of bands today. They are a self-contained band that writes its own songs and actually plays together. I like that. We recorded them live in the studio, keeping the bass, drums, and rhythm-guitar tracks. On one or two occasions, I actually used Donovan's scratch vocal on the final mix because it was so good.
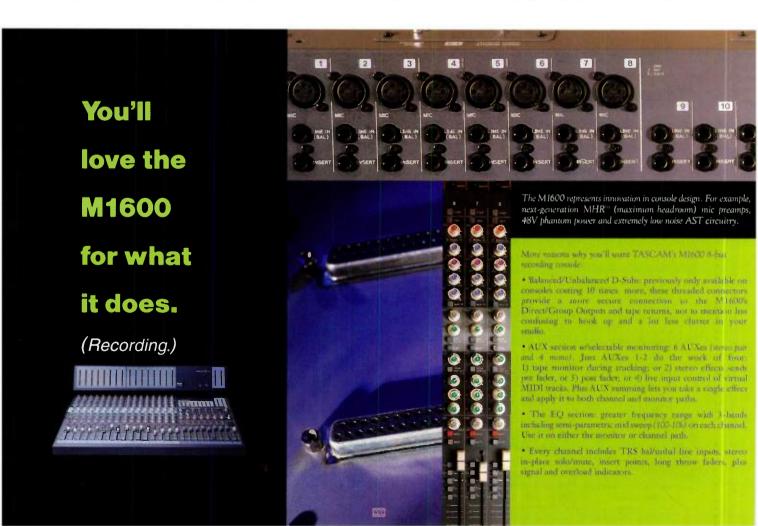
And this brings up another aspect of record production: There is no set way to do it. You try to plan for every situation and use the best of what happens. Sometimes the scratch vocal is the right performance. You've got to be able to hear that.

Speaking of creative hearing, your productions always have these wonderful ear-candy bits—such as the sparkling tremolo guitar that animates the choruses of the song "Colors" on Nancy Boy. *How do you come up with those things?*

I wish I knew how I figure out the parts I use to sweeten a track because I'd bottle it. The only thing I can tell you is that I get an idea for a part, and it just feels right. I'll hear a track and think, "Gee, this needs a bit of, um, washboard" or whatever. Again, back to the production-class thing, you can't teach a person to hear this stuff. It's not a science, it's an art.

Now, when you finally get to the mix stage, what is your primary goal?

The mix is the wrap-up. You're finishing up the story. It's like editing a film

where you have to put all the components together and come up with a box-office hit. This is the stage where you use every bit of experience and expertise to reach that point. It always turns out that the real crunch part of producing a project is the mix. The mix is where you make or break the record.

That statement about the mix making or breaking a record gets tossed around a lot, and I've always found it a vague and ominous threat. Specifically, how could someone kill a record in the mix?

For the sake of argument, let's say that you have a 24-track master on which each track was recorded extremely well. In the wrong hands—or with the wrong concept—those 24 tracks can be put together so that they sound like absolute crap. The music can turn out sounding unbalanced, sluggish, and without dimension or overtones. The EQ can be all wrong from top to bottom. Simply put, you can make a total hash out of it. Now, those same 24 tracks in the hands of somebody who *is* good at mixing can come out sounding brilliant.

TALMY'S GLAM BAM GUITARS

Shel Talmy's classic British productions tended to have real rough-and-tumble guitar tones. That tradition is maintained on the debut album by Nancy Boy, which showcases all kinds of glam guitar sounds with spanking mids and delicious distortion.

"We used some vintage gear for the guitar tracks," explains Talmy. "Jason [Nesmith, guitarist] ran through an old Matchless open-back combo with two 12-inch speakers and an old Marshall loaded with 30-watt Celestion speakers. I miked each cabinet with a Neumann U 87 set to a cardioid pattern and placed the mics three to six inches away from the speakers. On the Matchless, I also pointed a Sennheiser MD 441 dynamic mic into the open back and set that mic out of phase to the U 87. The resulting stereo picture was massive."
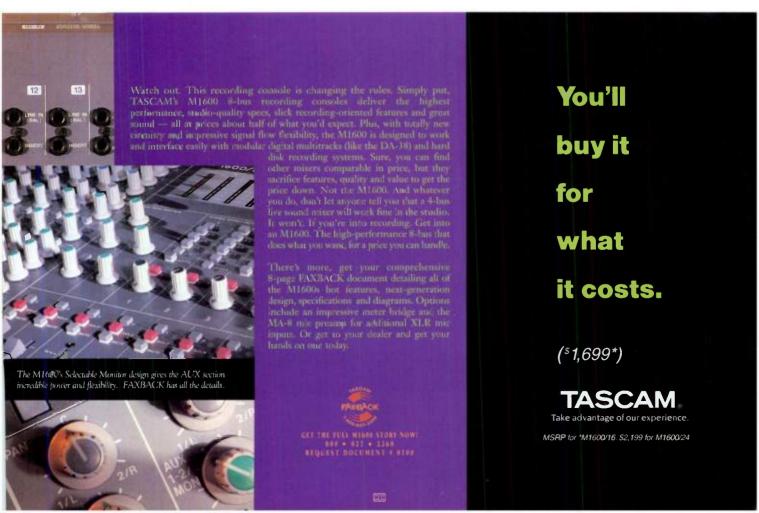
I must admit that I envy your working with bands such as The Who and The Kinks when they were young and full of passion. Those must have been exciting times.

Oh, yeah. Actually, the one thing I regret about those days is making the decision not to hang out socially with the bands. We were all roughly the same age, and I thought I'd have no authority in the studio if we were friends. I would only hang out in situations where, say, the Davies brothers were

kicking the heck out of each other, and the rest of the band and I would go and have tea until things settled down.

In the early days, most British studios were run quite formally; some even had rules for setting levels and equalization. Did you experience any culture shock?

I think the "shocks" were more on the British side. When I first got to England, most of the engineers were miking drums with four or five microphones,

and I started using eleven or twelve. They thought I was crazy because we were only recording on 3-track decks. I said, "That's okay. I don't care what you think. This is the way I'm going to do it."

I was also very concerned with sound isolation, so I'd spend hours and hours with an engineer building our own isolation booths and carpeted risers for the amps. No one else was doing this at that time. I was very hot on using isolation booths for vocalists, and in some cases, I was able to use the live vocal on the record because I had enough separation between the singer and the band. I was definitely trying to get as much down in one take as possible because the most I could hope to overdub without risking a ton of noise was a single bounce from one 3-track machine to another.

Recording sessions back then were unbelievably fast by today's standards; it wasn't unusual to cut an entire single in one day. Was the speed due to the recording budgets being so stingy that you had just enough time to get in and out?

Well, yeah, our recording budgets were pretty small, but that wasn't the only reason we made records so fast. People tend to forget that, back then, the bands' attitudes were different. It was uncool to take a lot of time in the studio because it meant that you sucked; you just weren't good enough to get in there and do it quickly.

Obviously, that attitude has taken a 180 degree turn, and it's entirely the opposite now. In those days, the better bands took *less* time to complete a record.

Today, everyone seems to be touting vintage recording gear. What was it like working in studios when that stuff was new?

There's no comparison. The tools today are five thousand times better. We were working with primitive equipment. But that doesn't mean I'll rely solely on modern technology to make a record

today. I'll use whatever sounds the best for a particular situation. Obviously, I still use some of the old tube gear because it sounds good. But for someone to say that they'll only use old stuff—or only use new stuff—is crazy.

However, I *do* believe analog tape is the right way to go for recording rock and roll. We tracked *Nancy Boy* on an analog Studer 24-track. I needed more tracks on a couple of songs, so I slaved a TASCAM DA-88 to the Studer. I'm happy about using digital—whether it's an MDM or [Digidesign's] Pro Tools— if it's warranted, but analog still sounds better for rock than digital.

So what does the future hold for you?

Well, I'm back producing full-time again, and it's nice to be somewhat in demand. The rumors of my demise were slightly exaggerated, so to speak. The problem these days, compared to the '60s and '70s, is that the lawyers have gotten into the act and things take five thousand times longer to sort out. I have a couple of projects lined up, but I have no idea when they're starting because the lawyers are still screwing

around with the contracts. I'm just going with the flow these days.

Youth (aka Martin Glover) pounded the bass for industrial punksters Killing Joke in the late 1970s and has since become one of England's hottest producers. The triple-threat producer-songwriter-musician has a discography chock full of productions and remixes for U2, Crowded House, INXS, Erasure, The Cult, James, Faith No More, Siouxsie and the Banshees, PM Dawn, and a horde of other modern-rock icons. He has also taken on a revitalized Vegas sexpot (Tom Jones), remixed a dead rock god (Jimi Hendrix), and updated his own past (producing *Pandemonium* in 1995 for the re-formed Killing Joke). The unrepentant workaholic also runs his own record label and studio complex, making it rather
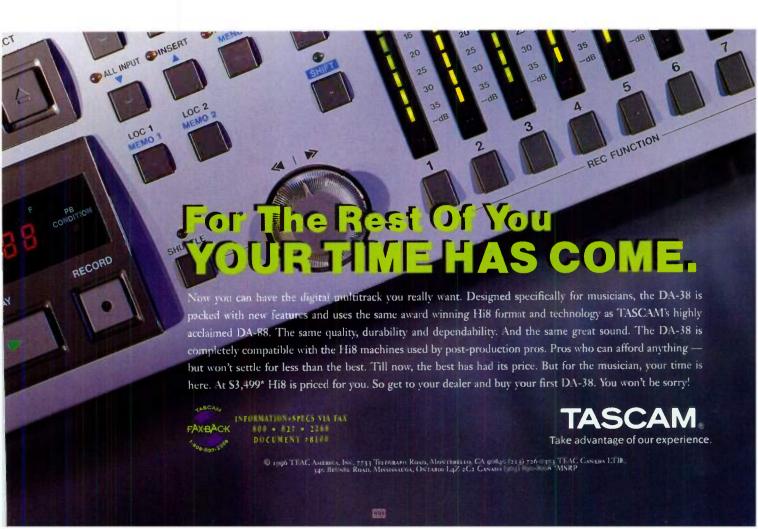
Youth (right) in his role as bassist-producer for the band Killing Joke.

easy to see the truth in Crowded House–vocalist Neil Finn's assessment that Youth is "a bit of a nut."

How do you conceptualize the role of the producer?

Well, I don't believe it works well to tell people what to do. The goal is not to enforce your ideas on the act; it's to realize and unify the shared vision of the band, the songwriter, and the producer. Unfortunately, some producers

don't even use the musicians who are in the band. For example, when I was working with Crowded House [on *Together Alone*], I was talking to the bassist about reviewing his bass lines, and he said, "You mean I'm going to be able to play on this album?" They weren't allowed to play on their own records; session musicians were brought in to play the parts.

It's okay to do that sometimes with singers and solo acts, but a band's bag is all about how the musicians play together. It's not about having a bunch of session musicians fill in; otherwise I might as well record the album by myself, and I'm not going to do that. I'd rather look at each musician's parts and try to make the best of them that I can. If a part really isn't right, I might suggest something else. More often than not, my suggestions are about stripping the part down and making it as simple as possible to cut out the clutter. I mean, at the end of the day, it doesn't really matter who's playing the part—a session musician or a band member—all that matters is that what goes on tape is right for the song. That's the goal I try to navigate toward. After all, record production is basically about producing a result.

But let's say you're producing singer-songwriters in the early stages of their careers. Even if you have to completely build the tracks from scratch for the artists, you still want to consolidate what they are and help realize *their* genius, not impose yours.
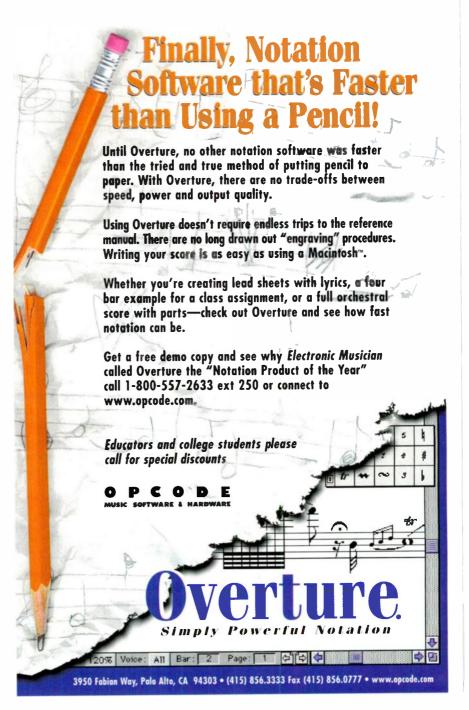
Because you are a songwriter and a musician, do you extend your production responsibilities to include editing and polishing an artist's songs?
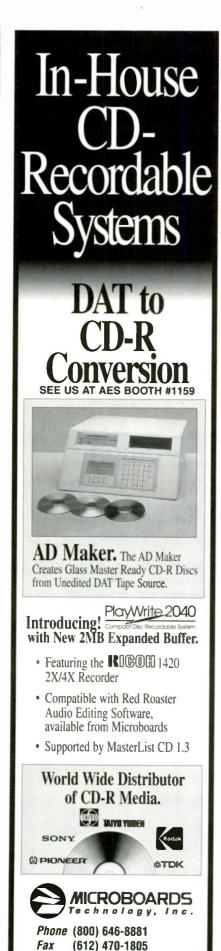
I do, but working with songs and songwriters is delicate stuff. Essentially, you can hear a good song with a voice and an acoustic guitar or piano. The song should shine with or without a full arrangement behind it. In fact, when I start the pre-production process, I usually get the band to sit around and listen to the vocalist singing the song in an extremely stripped-down version,

perhaps with just a single guitar as accompaniment. Such "unadorned" listening should give each person in the band an opportunity to better hear what the song is saying, without all the associations that different instruments bring to an arrangement.

Much of my role at this stage is to help the artist develop a vision for each song. Writers tend to get very close to the details of their songs and often cannot conceptualize an overall theme. I know this from writing my own music. You hear your work in a certain way,

but someone else can come in and take it further because they're in more of an objective position to see possibilities that you can't. So I listen to the artists' songs and sit down and discuss what each song means to them. Then I try to help them expand the songs to best achieve what they want to communicate.

Does this creative assistance include suggesting lyric changes?

Again, it's a delicate process. You might think a line is not right, but it's close to

Shop Talk

what the artist is trying to express. I still might try to encourage them to develop the line further. I'll just say, "I've got a problem with this line. Can you think of something else?" It's usually not a big deal. I mean, there are always lyric changes in the song right up to the time you cut the final vocal. Many of these changes are for meter—to help a line fall into the rhythm of the track better.

And I have to say here that, although I may suggest lyric changes, I never ask for songwriting credit unless I was initially commissioned to cowrite a project. That way, there is no confusion about your role, and you're free to give 100 percent as a mixer or a producer or whatever you were hired for. If the band wants to cut you in on the publishing, that's up to them. It gets too weird otherwise. I mean, you can literally "rewrite" parts of a song in the mixing process, but that doesn't mean you should get a songwriting credit.

It's interesting that you also produce techno tracks that typically do not follow standard song structures. Is it strange for you to be switching your brain back and forth between pop conventions and more unfettered musical styles?

If conventional songs are not what the band is about—say, it's an industrial or ambient act—the recording may come down to an emotional atmosphere. But generally, I still encourage people to stick to a standard song arrangement, no matter what type of

music they're doing, and then experiment *within* that form.

I've found that whenever you start veering off from the verse/chorus structure—which is basically an old folk-music arrangement—you confuse people. People are preconditioned to certain expectations when they hear a Western pop song, and if those expectations aren't delivered, they sometimes tune out. It's unfortunate, but it's true. Now, this is not necessarily the case with world or ambient music, but those genres still have recurring themes that *act* like pop choruses. Still, you must keep in mind that those musical styles have developed an entirely different language and people expect different things from that language; their preconceptions are different from those of a pop audience.

But we can't ignore that conventional song arrangements work really well. The format can be quite liberating, actually. Bands that want to do something totally unique are often locking themselves into a gilded cage. For one thing, anyone who thinks they are being "original" by having a strange arrangement is fooling themselves. Let's face it: it has probably been done already.

When you embrace the traditional arrangement, however, it leaves you the space to get into the parts you're playing and into what the song is really saying. In other words, you can discover the emotional content of your musicianship. You're not going through this intellectual process of thinking, "Oh, should we have no chorus on this song?" The real liberation comes from accepting the restrictions of the conventional song form because then you can sidestep the intellect and go straight for the emotional core. And that's exactly what you should be looking for as a musician *and* as a listener.

YOUTH'S MUSICAL AMNESIA

"One of the biggest things to remember when you're producing an act is to forget about the music," counsels Youth. "It's more important to create a vibe where the artist can have some fun. I mean, a guitarist may have studied his or her instrument for ten years and feel that making a record is the culmination of everything they've learned about the guitar. But actually, the record has nothing to do with the guitar. The guitar is just a vehicle for expressing the player's emotions. The music you make reflects what is going on inside of you, so if you're thinking exclusively about the technicalities of performing music, you simply will not have much to say."

You should strive to put something into a song that is so uniquely yours that no one can do it, except for you. Now, *that's* being truly original.

How exactly do you inspire a musician to seek out his or her "emotional core"?

When I've produced ambient or trance projects, visual metaphors have been very helpful. I might have the artist imagine that we're in the foothills of the Himalayas trying to get to this valley. So I'll then ask the artist to envision the first part of the song as a journey where we are rummaging around the undergrowth of these foothills. The metaphors provide a navigational plot, but they also give the artist complete creative freedom.

Metaphors can work for individual parts, too. If you're a guitarist, I might tell you that you're a wounded buffalo being chased by Indians. You're stuck in some mud trying desperately to get out, and they're shooting arrows at you.

Now, I want you to make the guitar sound like what you feel.

What I try to do is to sidetrack the intellectual, conscious mind that tells you, "I have to play all these notes in this particular sequence, and it has to sound like a specific personal ideal." For example, one bass player I worked with had a lot of trouble doing his overdubs. Every time we'd go to record, he'd get what we call "red-light fever." Whenever the record light went on, he'd freak out. I found that he was concentrating too hard on the parts, so I would use these little tricks to get him to think about anything *but* what he was playing.

Once, I had the tape operator read poetry into his ear to distract him from what he was playing. When I did stuff like that, we'd get a take in a couple of hours whereas doing it the normal way would take eight hours or more.

That's the test of a good producer: slog it out until you get what you need. You must be able to stretch your imagination to find little tricks and techniques that can help the musicians really shine. It's not about getting them to bend to work *your* way; it's about you bending to the way they are and helping them express themselves.

So much of production is psychology, really, rather than actually knowing how to use this or that piece of gear. You have to create a nurturing environment in a cold studio crammed with cords and wires and high-tech equipment and all these engineers running around. The production process is fraught with paradox, and that's the beauty of it. You have to embrace various dichotomies: technique versus passion, intellect versus emotion, and so on. But you can't have one without the other. It's like a musician learning scales just so he or she can forget them. You have to practice certain things so you can perform without thinking about it.

Do musicians react favorably when you steer them into uncharted territory? I know that many players are particularly uptight about having to surrender technique for passion.

The great thing about producing music is that you're working with people, and everybody is different. And even within that sweeping diversity, individuals act differently at different times of their lives, so there is never any guarantee how someone is going to react to your suggestions. You must be quite flexible in how you approach artists and, at the same time, pretty firm about where

FIVE EASY PIECES

So you want to be a record producer, eh? Well, getting your production chops together in front of a band and an engineer or two can be a scary and humbling experience. To help you survive some very difficult situations, I'd like to share five tenets that kept me reasonably sane throughout the panic and paranoia of my first production gigs. Good luck. And remember: you can always erase your mistakes.

1. Your name is not Adolf (or Phil Spector). There is no rule that says a producer must be a dictator. Repeat after me: the recording process is a collaborative experience. You're there to help the artists maximize their gifts, not to take control of the circus and make everyone so miserable that, ultimately, they can't bear to listen to their own record.

2. It's not your album. The artist is hiring you. Do whatever you can to ensure that the best stuff gets on tape, but if the artist is adamant about doing something you're not thrilled with, just shrug your shoulders and move on. (You can always bury less-than-brilliant parts in the final mix.) For better or worse, it's the artist's record.

3. Don't do something just to do *something*. There will be times when the artist doesn't need your help. A brilliant song or performance will simply spring onto tape fully formed and exquisitely faultless. Do not soil perfection just so you can feel that you've contributed something. Shut up and enjoy the moment.

4. The producer's job description includes stuff that is absolutely terrifying. Don't think for a moment that your production duties end at evaluating performances and suggesting cool parts. No way. You are going to be shackled to the artist's psyche for the length of the recording process. You will be this person's therapist, cheerleader, teacher, roadie, social worker, and surrogate daddykins or mommy dearest. (And producing bands is even more fun because each of the band members will want his or her share of your attention.) Deal with it.

5. Let the playback be the "bad cop." A revealing playback is worth a thousand arguments. Don't get an ulcer—or even waste valuable time—trying to convince a brutish singer that he or she is too tired to deliver a good performance or reasoning with a rhythm section that can't seem to play together. Just press Play. When the artists hear for themselves how bad they sound, their sense of self-preservation will prompt them to see things your way.—*Michael Molenda*

you want to take them. This is a great challenge because I don't believe in compromises in the studio. Ideally, I believe that everybody has to get what they want out of the recording.

To do this with any hope of success, you must sort everything out in the project's initial stages to ensure that everybody is making the same record. You must take a bit of time away from the details of making a record and discuss the bigger picture.

Quite a lot of producers forget this stage, and then halfway through the album, they discover that they're making a completely different record than the bass player is. Or that the vocalist is making a different record than the guitarist is. Of course, you have to leave a certain amount of room for spontaneity, but working out a good navigational plot for a record makes it a hell of a lot easier to get to the common destination because you know where you're going before you leave.

ALL TALKED OUT

As you've probably surmised by now, there are a number of different methods and techniques that producers employ to make a record. Unfortunately—for those who need a clear barometer of right and wrong—there are no hard-and-fast rules for success.

Having said that, two ideals resonated through the words of Seibold, Talmy, and Youth that may give you a fighting chance at producing good work. First, each producer, in his own way, possesses the imagination and determination to get the right stuff on tape. Second, each producer respects the artist's vision.
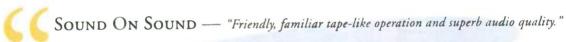
Believe me, you could do much worse than starting your own list of production ideals with the words *determination* and *respect*. Now, get out there and make some records!

EM *Editor in Chief* **Michael Molenda** *is currently producing albums for Smitty, Eva Jay Fortune, Ascot Jacket, and Soul Poets. He is also wondering where his social life went.*

ALBINO APEMAN CAPTURED BY EXPLORERS!
ProR3 Reverberator Used to Coax Wild-Eyed Yeti From His Himalayan Hideaway.

Half man, half ape, the legendary Yeti has roamed the snowswept peaks of Nepal undisturbed for centuries. Only the most courageous mountain climbers and a few indigenous natives have even come close to sighting the standoffish beast, characterized by a powerful odor and size-20 footprints. Known in other parts of the world as Bigfoot, Sasquatch and the Abominable Snowman, the Yeti has yet to be captured... Until now!

Pro R3

"We had a devil of a time tracking this chap," declared exhibition leader Sir Arthur Mildmay of Medford, England, who was on assignment from the Queen herself. "It wasn't until we hooked the Yamaha ProR3

Digital Reverberator up to the vocals of our yodeling colleague, Mr. Friedolfson, that we were able to duplicate the exact resonance of the Yeti mating call." Too bad they couldn't use the unit to deflect the abominable smell of that putrid primate!

From the same people who created the 02R, the $1599 Yamaha ProR3 Digital Reverb proved its worth in the field. Comparable to units that cost thousands more, the ProR3 runs on a new, powerful DSP engine that allows higher reverb density and the ultra-smooth decay, just the ticket for professional studio production or for attracting an arduous ape in the wilderness. Armed with 32-bit digital sound processing power and high performance 20-bit A/D and D/A converters, Team Yeti was able to

produce unprecedented reverb effects needed to lure the beast.

"We experimented with the 90 ProR3 presets and 90 user programs until we came up with the right combination of reverb, delay, early reflections, gating and EQ," said Mildmay. "It seems the Yeti was seduced by an exceptionally smooth and thick reverb sound, like in the stone wine cellars of my home town."

In his new home in the Land's End Zoo, the ProR3 is used to coax Yeti into believing his 12 foot square cage is actually the size of Mt. Everest. "The ProR3 is quite effective. Still, it's obvious he's not exactly the brightest fellow," said Mildmay.

his inter-planetary voyage—an otherworldly assortment of musical instruments, including a "trumpet" with a 500 mile long flute shaft and a clarinet bell. "It seems like the only people on Earth who can hear these instruments are my dog and some cockroaches," he explained, "but the babes on Saturn dug 'em."

Ranger and the reptilian-looking alien group "Earthworms" created the instruments using the Yamaha VL1m and VL70m tone generators and VL editing software. "We're zipping past Mars and the Earthworms say my instruments are 'too terrestrial.' So they download Macintosh software from www.midifarm.com/yamaha using this 40 billion baud modem."

Next, Ranger reports, the Earthworm's manager, Xeobojb, used the Visual Editor to mix and match instrument parts like reed type, pipe type and a basic instrument shape. Then he edited and tweaked various envelopes and controls. With the Analog editor he adjusted pitch, vibrato, attack, decay, etc. "Zeeb made some awesome stuff. It helped that he was pretty high. At least I think he was."

The Inquisitioner has verified Ranger's claim that VL1m and VL70m editors for the Macintosh and Windows do actually exist at www.midifarm.com/yamaha, though not all editors are available for the Windows platform.

Those in search of extraterrestrial life take note: Ranger promises that the Earthworms will return to Rantoul next Friday to pick him up. "We've got another gig out near the Ring Nebula. I'm not too crazy about the music—it's not rock 'n roll—but a gig's a gig."

VL70m

ANIMALS SING TOGETHER TO SAVE ENVIRONMENT

"Goats don't particularly like lions; they tend to get eaten by them," explains Zed Mundo, who overcame this and many other roadblocks to produce the hit song "We Are The Animals," completely sung by members of the animal kingdom. "I got a coyote and a rabbit in the same room to sing a duet and it got pretty hairy. That's when I knew I had to figure out a different way," he said.

The song's co-writer, Franco Ziff, suggested the answer. "I advised him to try the new Yamaha MD4 digital four track recorder. At just $1199 the MD4 is the least expensive digital

multitrack, and that's the limit of what we could afford. Sonically, it's vastly superior to cassette multitracks."

Mundo carefully wove the dynamic song's fabric together on the MD4. He mixed and edited 790 different animals' vocals and bounced them from track to track with zero loss of sonic quality. "Working with the MD4, it's like I have more than four tracks at any given time because I can combine completed tracks onto any track—even if there's audio on it—without losing that track. Also, I didn't need to save a track for MIDI timecode. The MD4 sends that out separately," Mundo reported.

The MD4's MIDI timecode was a saving grace as Ziff composed the music's haunting theme on a Yamaha QY700 using 80 tracks of his own.

Bill Dider, an engineer at Banana Records, was impressed with

the recording quality. "When I heard it was coming in off a portable multitrack, I thought we'd have to redo the whole shebang. But we're talking studio quality here. Flat frequency response from 20 Hz-20 kHz, 44.1 sampling rate and there's no audible distortion. And I never could tell where he made his edits thanks to the MD4's precision editing functions. Good thing I wasn't looking forward to re-recording that rattlesnake."

The song has so far raised $5 million to help the environment. We may do a follow up," Mundo said, "but with no monkey this time. What a prima donna."

MD4

The
Magnificent Seven

It all started with one or two tough customers preparing to take on the bad guys. But slowly, one by one, like-minded companies entered the picture with new products, until seven heroes stood ready to fight in the great Format Wars.

Yes, every time you turn around, another manufacturer conjures up a new 8-track modular hard-disk recorder (M-HDR). Today, seven magnificent 8-track models are battling

Seven amazing

8-track hard-disk

recorders ride

bravely to the

Format Wars.

By Erik Hawkins

for market share. Akai's DR8, E-mu's Darwin, Fostex's DMT-8 and D-80, Roland's DM-800 and VS-880, and Vestax's HDR-8 are all excellent machines that could completely change and radically improve the way you record music.

The battle for market dominance began slowly. In 1990, Roland released the DM-80, a big beast that worked great but cost approximately $10,000 for a full-blown system. Two years later, Akai released the first affordable M-HDR, the 4-track DR4d. Although these machines gained some popularity and introduced many people to the virtues of random-access recording (e.g., no rewind or fast-forward time, multiple levels of undo, and cut-and-paste editing), storage and archiving was a real pain. For a short while, the two champions stood alone.

Fortunately, the technology has advanced considerably since the introduction of the DM-80 and DR4d. The advent of seven magnificent 8-track machines appears to be a harbinger of changes to come. (We're only comparing 8-track machines here, but two other M-HDRs deserve mention. Akai's DR16 is a 16-track version of the DR8, and Vestax's HDR-6 is essentially a 6-track version of the HDR-8. In addition, at press time Vestax announced the HDR-V8, a version of the HDR-8 that will have eight analog inputs, eight analog outputs, an optional SyQuest SyJet 1.3 GB removable drive, and optional TASCAM and Alesis interfaces.) Today, the modular HDR is moving firmly into all kinds of studios. Let's watch these powerful champions ride into town and see which is your best choice.

THE MOST RELIABLE ALLY

Why should you buy a stand-alone HDR? After all, a computer-based, multi-track digital audio workstation (DAW) offers the same features and more, including graphical waveform editing on a computer monitor. And the DSP plug-ins for some computer systems have significant technological advantages over hardware effects processors.

Well, for one thing, computers have a tendency to crash. If all your MIDI sequences and audio tracks are in your computer, you're in big trouble if your system goes down. I've used almost every M-HDR released since the DM-80, and I've never had one crash. This is pretty amazing, considering how many lost and corrupted tracks I've had because a DAW decided to go on sabbatical in the middle of a recording session. And the harder you push a computer-based system by running more DSP software and recording lots of audio tracks, the more RAM and CPU speed you need, making a system crash more likely.

In addition, modular HDRs are generally less expensive and more portable than computer-based DAWs. Sure, you can run a digital audio sequencer on a Power Mac PowerBook with Sound Manager audio or a Windows notebook computer with a sound card. But a notebook computer–based system that can deliver comparable recording performance to an M-HDR is relatively expensive, and you have to edit on a small screen. Furthermore, most M-HDRs have built-in mixers with plenty of inputs and outputs and high-resolution digital converters, which you don't get on a notebook computer.

Most M-HDR manufacturers realize that technically savvy musicians have become accustomed to the advantages of graphic waveform editing. Roland's DM-800 and Akai's DR8 have monitor outputs for track and waveform viewing and editing, but they don't have the same feel as a DAW (although the DR8 comes close).

One solution is to use a DAW for waveform editing, some types of effects processing, and CD-R mastering while using the modular HDR for composing, tracking, and mixing or submixing.

The machine with the most potential for seamless integration with a DAW is E-mu's Darwin. Darwin stores its audio data as WAV files and formats its disks as Windows volumes, making its files readable by a Windows PC. Imagine pulling a removable disk out of a Darwin equipped with an onboard Iomega Jaz drive, sticking it into your PC's Jaz drive, editing the tracks, and then sticking it back into your Darwin. At the moment, Darwin's files can be read by a PC running wave-editing software, but Darwin cannot read WAV files coming from the PC, including its own edited tracks. E-mu hopes to make this exchange a 2-way street in the not-too-distant future.

DON'T GET TRAPPED!

The number and type of user-accessible expansion ports often dictate an M-HDR's life expectancy. It is important to have a unit that can grow with you. Expansion ports will allow you to add SMPTE time-code support when you get that big film job, or sync to an MDM tape recorder when you expand your studio, or add storage when you can afford it. Without expandability, you might eventually have a severe case of buyer's remorse.

E-mu's Darwin, Vestax's HDR-8, and Akai's DR8 have several expansion ports. These slots, when filled, yield connections to the outside world, such as additional audio inputs and outputs and interfaces with other multitrack recorders.

The DR8 and Darwin have the most external expansion ports, so there's plenty of room to grow with either machine. The DR8's five large ports can admit SCSI or some other multipin connector. Two smaller ports combine to admit a card with a PS 2–style keyboard connector and a Superview monitor output. The Darwin also has an internal port that can be used for the new DSP expansion board. Roland's VS-880 only has one expansion port. It is internal and is designed for adding an effects processor; it cannot be used to provide connections to the outside world.

The Fostex machines and Roland's DM-800 do not have expansion ports.

The Akai DR8 is extremely well endowed with pro features, including expandability, plenty of inputs and outputs, and SMPTE sync capabilities. Its main drawback is that it can't sync directly with the ADAT and DA-88.

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However, the DM-800 already includes most if not all of the features that you could add via expansion boards.

At the moment, Akai is way ahead of everybody else in expansion-card development. Vestax is right behind, but no matter how much effort Vestax puts into development, with only two expansion ports on the HDR-8, it is likely to come up short in this department compared to the Akai and E-mu machines. The Darwin has several expansion cards already, despite its fairly recent arrival on the scene. (For a complete listing of expansion cards, check out the table "Expanded Armory.")

PROTECTING YOUR REAR

An M-HDR should have as many storage options as possible. Clearly, the best solution with the current technology is to have a SCSI port. With a properly implemented SCSI chain, the data can stream to the next attached SCSI drive if your song is longer than one disk can hold. In addition, SCSI makes it easy to back up data directly to fixed or removable media.

The DR8, Darwin, VS-880, and DM-800 can record directly to a SCSI disk in real time and can back up via SCSI. The Vestax only uses its SCSI port for backup. Neither Fostex machine supports external SCSI devices.

Iomega's 1 GB Jaz drive is one of the most attractive SCSI-based removable media. The VS-880 and Darwin can accept internal Jaz drives, so these units are poised to reap the maximum benefits the Jaz system offers. In the near future, I expect to see a lot more HDRs with onboard Jaz drives. However, at about $120 each, Jaz disks are expensive. If you can't afford a Jaz drive or extra Jaz disks, it's advisable to make sure you can back up to a hard drive or audio DAT.

The Fostex machines provide backup to audio DAT. The Fostex D-80 uses an IDE hard disk encased in a removable cartridge, which provides additional recording time by allowing you to swap drives. This is nearly as simple as swapping a Jaz or SyQuest cartridge, though you have to unlock the drive using a small key. Supplemental drives and cases cost almost as much as Jaz cartridges. Because the unit can't connect to a second drive, it doesn't provide overflow for extended, contiguous recording. To use it for backup, you have to back up to audio DAT and then restore to a different cartridge.

The DMT-8 can back up two tracks at a time to audio DAT, but the process is relatively slow (four times real time). However, DMT-8s with OS 2.0 or higher let you partition the hard drive so you can put a project on its own partition. That lets you back up a single project quickly without touching other projects.

Both Roland 8-track machines, the Vestax HDR-8, and the Akai DR8 support audio DAT backup. Darwin does not back up to audio DAT because E-mu feels that audio DAT backup is less reliable than SCSI backup.

The DR8 also has optional backup

This **TD-5K Compact Drum System** comes with a TD-5 sound module, five PD-5 pads, a KD-5 kick trigger, an FD-7 hi-hat controller, stand and cables. With 210 dynamic, 16-bit CD-quality sounds and a digital effects processor, this kit offers the perfect combination of power and affordability.

The FD-7 hi-hat pedal gives you all the control of a real hi-hat: pedal, heel-splash, open to closed and everything in between.

Roland's PD-7 and PD-9 dual trigger pads feature rim shot and cymbal choke capability.

The **TDE-7K Compact Drum System** comes complete with a TD-7 sound module, eight TD-7 dual trigger pads, a KD-7 kick trigger, an FD-7 hi-hat controller, stand and cables. For the ultimate in flexibility, the TD-7 sound module has 512 16-bit CD-quality sounds, an onboard digital effects processor plus a built-in sequencer.

to ADAT via Alesis' optical connector; the other manufacturers have not yet caught on to this idea. With the ADAT Lightpipe expansion board, it is possible to simply transfer your tracks in real time or to back up your tracks, edits, locate points, and anything else associated with a project directly to ADAT. If you already have an ADAT, this is a quick and inexpensive back-up method. You can easily fit 1 GB of information on a $10 S-VHS tape.

GETTING IN AND OUT

From live recording to studio mixdown, you need individual analog audio inputs and outputs on all tracks to get the job done in a professional manner. Without a full complement of individual track outputs, you cannot route individual M-HDR tracks through an external mixer. This limits both the type of outboard effects you can apply and the timbral qualities of your final mix. (Mixing tracks through a classic British console and a high-end reverb sounds a lot different than mixing with an M-HDR's onboard mixer and internal effects.) Without a full complement

The E-mu Darwin may be the prototype M-HDR of the future. It is affordable, offers a great user interface, has lots of ins and outs, is expandable, and boasts an excellent implementation of virtual tracks. However, the Darwin's mixer lacks EQ and can't be automated via MIDI Control Changes.

of individual inputs, you can't record on all tracks simultaneously, which is a big problem for live recording.

The DR8 and D-80 are the only machines with a full set of eight track inputs and eight track outputs. The DR8 and D-80 have these ports as standard equipment. The stock Darwin has eight track outputs and only four track inputs, but four more inputs can be added with the Model 4010 expansion

card. All DR8 and Darwin connections are on ¼-inch balanced jacks; the D-80 uses RCA connectors.

The DM-800 has four balanced analog outputs that can be assigned to deliver any direct track output, the main L/R mix, or the unit's internal mixdown bus. The latter is essentially a stereo aux send that can carry the L/R mix or any pre- or postfader track output. Outputs 1 and 2 default to the L/R
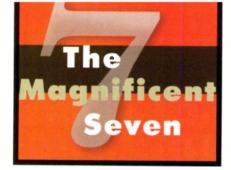
mix. In addition, you can add the DA-400 output expander, which gives you eight analog outputs.

Except for the Darwin and the D-80, every M-HDR I examined has a set of analog master L/R mix outputs, which are meant to be used with the M-HDR's onboard mixer. The Darwin only has digital L/R mix outputs, which I'll discuss later. The D-80 is excused because it doesn't have a mixer.

Being able to mix your tracks within your M-HDR and output them via the stereo mix outputs is an expedient and space-efficient alternative to using the usual large mixing board with cables running all over the place. Even if you don't want to do your final mixes in this fashion, these outputs are great to have on hand for submixing and auditioning rough mixes.
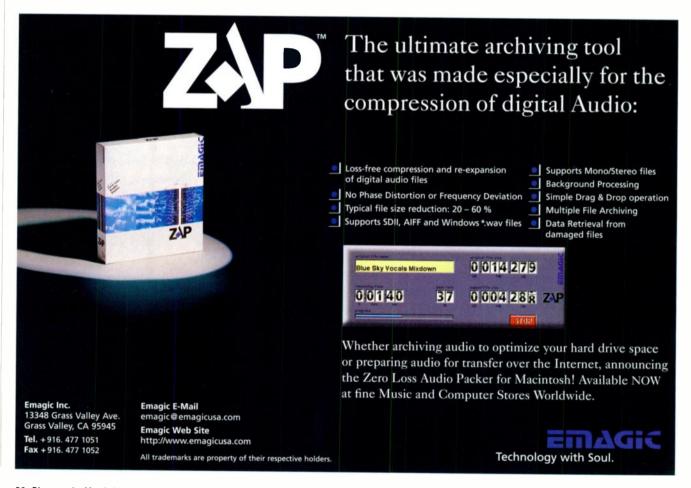
Fostex's DMT-8 is clearly aimed at the cassette Multitracker user. It's the only M-HDR with an analog mixer, and it lacks many professional features. But it sounds very good, and it is the least-expensive 8-track M-HDR.

The M-HDRs with onboard mixers also have at least one set of stereo aux sends for hooking up an external effects processor. The Vestax HDR-8 has two sets of stereo aux returns, as well. During mixdown, these ports can be used either as effects returns or as inputs for additional sound sources.

Every machine except Fostex's D-80 has a headphone jack and associated

headphone volume control somewhere on its face. (The D-80 has no mixer, so a headphone jack is unnecessary.) The Darwin's headphone amplifier seems very weak; I could never get enough "bump" out of it. The other units' headphone amps performed well.

MODERN WEAPONRY

Digital inputs and outputs allow a wider range of connectivity with DAT decks, computer-based DAWs, digital video decks, digital audio converters, and so on. In general, the more digital I/O, the better, but the important consideration is to get the right digital I/O for your needs.

For example, all M-HDRs have S/PDIF on RCA (coaxial) jacks except for the Fostex machines, which have optical S/PDIF connectors. A lot of digital gear does not have optical connectors, so if

you wanted to back up to, say, a TAS-CAM DA-20 or Panasonic SV-3700 DAT deck, you'd have to get a converter box to change the optical connectors to RCA jacks. Fortunately, Fostex makes an optical-to-coaxial S/PDIF converter box for only $95. If you use a TAS-CAM DA-30 mkII or Fostex D-5 DAT, which have optical S/PDIF, you'll be fine with the Fostex implementation.

The Vestax HDR-8 has S/PDIF I/O on both optical and RCA jacks, so you're covered either way. Roland's DM-800 has two simultaneously available S/PDIF outputs and one S/PDIF input on RCA jacks, so you get four digital output channels and two input channels.

The DR8 is the only machine that includes AES/EBU input and output on XLRs. E-mu's Darwin has AES/EBU I/O, but it is on the same RCA jacks as the S/PDIF I/O, and the two formats are not simultaneously available. The AES/EBU standard calls for XLR connectors, so you have to use an adapter to connect the Darwin's AES/EBU jacks to another device.

The Magnificent 7 Seven

The Fostex D-80 has no mix outs, so it allows any two tracks to be assigned to its digital output. The digital outputs on the other machines I tested act as mix outputs, mirroring the L/R analog mix outputs. Roland's DM-800, with its four digital outputs, allows any track or the L/R mix to be assigned to any digital output.

All seven M-HDRs allow the digital inputs to be assigned to any track. In all cases, the left and the right inputs can be treated as discrete channels and assigned to nonadjacent tracks.

The Darwin, DM-800, and DR8 can also be equipped with Alesis' Lightpipe 8-track digital interface, which allows you to transfer up to eight tracks back and forth between the M-HDR and an ADAT. For example, I like to lock my ADATs up with an M-HDR, transfer reference music tracks over to the HDR, and then turn the ADATs off and just work on the HDR. I can cut vocals faster and more efficiently this way, taking full advantage of random-access recording without fearing that I'm going to snap a tape from too many takes. After I have compiled my final takes, I can transfer them back to the ADATs for easy storage and mixdown. The Lightpipe interface lets me do all this without any signal degradation because all my track transfers are done completely in the digital domain.

Roland's DM-800 is the only unit that offers optional 8-track digital interfaces for both Alesis ADATs and TASCAM DA-88s (and DA-38s) and includes an RS-422 port. This requires Roland's DIF-800 breakout box, which connects to the DM-800's rear-panel RMDB 8-track digital interface. Akai has announced a DA-88 interface for the DR8, which should be available by the end of this year.

ADVANCING AND RETREATING
All of the units I tested have the familiar REW, FF, Stop, and Play transport buttons. However, because these machines are random access and have their own logic boards, a plethora of cue functions and locate points are

FOLIO (SX)

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- **Custom-designed consistent controls** give an even spread of control around their sweeps.
- **Surface-mount technology**
- **Rack mount option**

SX's Direct Outs on the first 8 inputs are from panel switchable pre-fader for live track laying or post-fader so you can "massage" recording levels in the studio.

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UltraMic™ padless preamp gives 60dB of gain range with 22dBu of headroom

Insert point for effects

Direct Out on channels 1 – 8

100Hz steep slope High Pass Filter

3-Band "British" EQ with swept mid

3 Auxiliaries configurable 2 x Pre/1 x Post or 1 x Pre/2 x Post

Channel "ON" switch

MIX/SUB routing switch

Full 100mm fader

PFL

STEREO INPUTS

13, 14, 17 & 18: simple phono inputs with level and MIX/SUB routing

15, 16, 19 & 20: Balanced jack inputs with Gain, 2 band EQ, Aux levels, Channel "ON", MIX/SUB routing, 100mm fader and PFL

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By Soundcraft (SX)

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STUDIO RECORDING
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SUBMIXING

LOCATION SOUND

TYPICAL SPECIFICATIONS

Mic EIN	-129dBu
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Channel Mute	<95dB
Fader Cutoff	<90dB
Frequency Response (20Hz to 30kHz)	<1dB
THD	<0.006%

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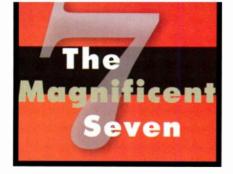
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The Magnificent 7 Seven

Roland's DM-800 sounds terrific, has a monitor output and a physical mixing surface, and supports direct sync with both ADAT and DA-88. However, this pro unit is by far the most expensive M-HDR.
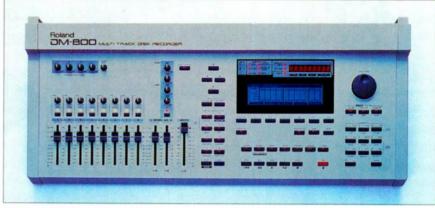
possible. This feature gives you the ability to audition anything and get anywhere almost instantaneously. The distinction between locate points, edit points, cue functions, and markers can get a bit murky, so before we compare the various implementations of these features, let's get our terms straight.

Locate points are spots that can be marked and directly accessed via a 10-key pad or dedicated buttons. Most machines have edit points in addition to locate points. Edit points are really special locate points that are configured as an In point and an Out point (also referred to as A and B points). A regular locate point can also be used as an edit point.

Cue (or preview) functions are specialized transport features used for auditioning edit points and are triggered with dedicated buttons. There are four basic types of cue functions: To, From, Between, and Over (or Thru). The To function begins playback from a user-programmable time before the edit point (a preroll) and stops at the Edit point. The From function plays from the edit point to a programmable time after the Edit point (a postroll) and stops playback when that postroll time has been reached. The Between function plays the region between two edit points, typically starting at the In point and stopping at the Out point. Over gives you both a preroll and a postroll,

so playback starts before the edit point and continues until it reaches a programmable time after the edit point.

In addition, most machines can cue to the last edit point that was selected. However, only the DR8 has a Last cue function, which can locate to the last two points where you stopped the machine, without the need to set a locate point.

Roland's markers are essentially locate points with no direct-access buttons. Instead, you use increment/decrement buttons to scroll forward or backward between the markers, one at a time. Both Roland machines use this system.

The Akai DR8 has the most cue functions on dedicated buttons: To, Between, From, Over, and Last. It has the usual edit In and Out points, nine locate points that can be directly summoned from the 10-key pad, and 100 Stack points that are almost as directly accessed: you just hit the Stack button before punching in the double-

digit Stack-point number with the 10-key pad.
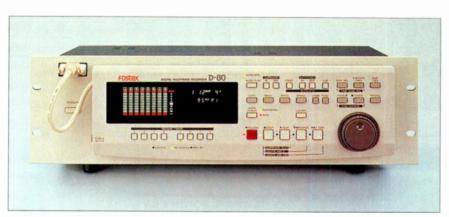
The Darwin has To, From, and Between cue buttons. However, the cue functions are difficult to access because they use soft buttons and are buried several pages beneath the main window. This problem can be somewhat ameliorated by the two assignable front-panel keys, which you can use for cue functions.

The Darwin also has 40 locate points (not counting the edit In and Out points), any ten of which can be accessed via the 10-key pad. The machine's big LCD screen shows ten locate points at a time with the associated SMPTE times for each.
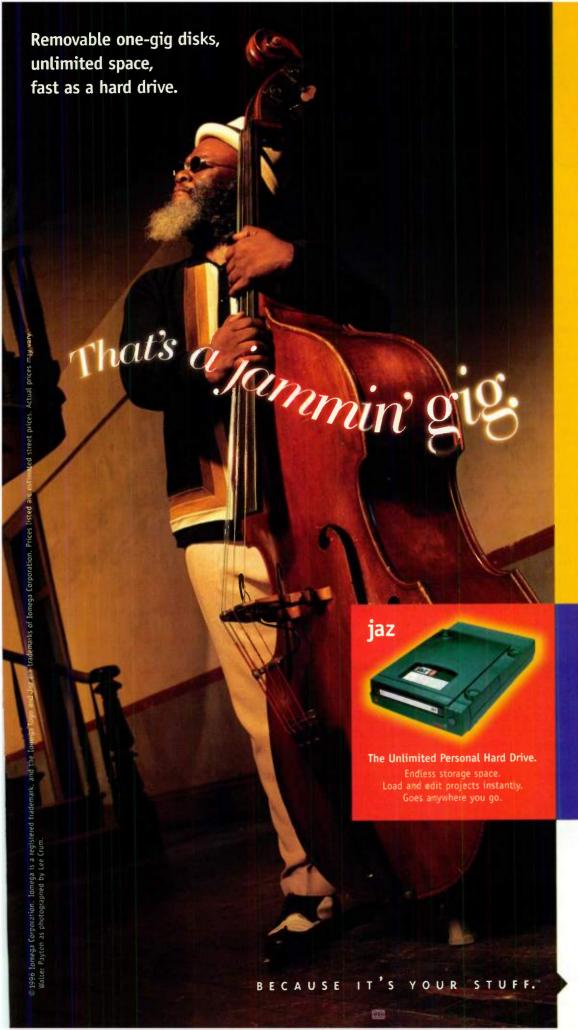
Roland's VS-880 is the most gifted when it comes to locate points. The original unit, which I had for review, had eight locate points on dedicated buttons and 999 markers you could scroll to with the left/right arrow keys, for a total of up to 1,007 locate points (plus edit In and Out points) in a single project. The latest update brings the number of locate points to 32, for a total of 1,031 points. In addition, the original unit's only cue functions were To and From; the upgrade adds a Thru function.

The DM-800 has dedicated buttons for three cue functions: To, From, and Thru (Between). It supports up to 40 markers, which are configured as five groups of eight points that can be assigned to eight dedicated buttons.

Fostex's D-80 and DMT-8 have the fewest features in this department. Each machine has eight locate points that are similar to Roland markers in that one button accesses all eight of the points, so you have to select which

Whereas other M-HDRs purport to be complete portable studios, the Fostex D-80 operates like a traditional tape deck with individual track inputs and outputs but no mixer. Unique to the D-80 are a removable remote control and a removable cartridge-encased hard drive.

locate point the button will access each time. The Fostex machines are also the only M-HDRs that lack dedicated cue buttons.

The Vestax HDR-8 has a solid if unspectacular implementation, providing eight dedicated locate points in addition to the usual edit In and Out points (labeled A and B). It also has To, From, and Between cue functions.

The Akai, E-mu, and Roland machines can jump directly to specific time-code locations via their 10-key pads. I wish the other manufacturers would adopt this useful feature.

Every M-HDR has a punch in/out footswitch jack. Every unit except the Darwin has an auto-punch function, which I consider extremely important. I count heavily on auto-punch when I am composing and recording alone. True, you can perform an auto-punch on a Darwin using MIDI Machine Control (MMC), but this isn't the same as having the feature available on the face of the unit. Besides, MMC is so slow that the punch-in point you designate isn't quite where Darwin ends up punching. Fortunately, E-mu plans to add auto-punch in a November firmware upgrade.

MONITORING THE SITUATION

All machines except the Roland VS-880 have bright, easy-to-see, multisegment, multicolored track-metering LEDs. The VS-880 loses out in a big way here because its LCD screen is smaller than a business card and is not backlit; yet it displays everything from track levels to SMPTE time and operating status. I could read the display, but it caused eyestrain, and I felt the other M-HDRs did a better job of displaying the track levels.

Every unit except the Darwin and D-80 lets you see not only the track levels but also the master L/R mix levels and the aux send/return levels. (The Darwin and D-80 don't have aux buses or analog L/R mix outs to meter. The Darwin only has digital L/R mix outs, and the D-80 doesn't have a mixer.) The units with internal mixers allow levels to be metered pre- or postfader.

How much information you can absorb in a glance is extremely important. A well-designed front panel should tell you the machine's status, the SMPTE time, the levels, and so on, quickly and without a struggle. The Akai DR8's front panel presents the most information at once, but E-mu's Darwin presents the most information in the clearest fashion.

The DR8's myriad discrete buttons and closely packed track meters clutter the front panel, and I found it hard to focus on what I wanted. The Darwin has a much sparser front panel that can display just as much information.

Instead of feeding all the information to you at once, its track meters just monitor levels; a separate, large LCD screen lets you scroll through the features and only display what you need when you need it. At different times, in different windows, the Darwin LCD screen can display everything from track names to location points to block diagrams of recorded areas on your tracks.

Not to be outdone, Akai recently released the Superview expansion card, which lets you plug the DR8 directly into a computer monitor. Ultimately, the ability to plug directly into a monitor yields the best user interface. It lets the user see what's going on via graphic representations in a space far larger than any LCD screen you could fit on an M-HDR.

Roland's DM-800 monitor output is a stock feature that gives you multisegment, multicolored track meters. Its output looks like something from an old Apple II computer—it's not as glamorous as the DR8's monitor out, but it gets the job done. The Darwin has a host-computer port for use in the future when editing software is developed, but the port does not connect directly to a monitor.

TACTICAL MANEUVERS

Increment/decrement buttons are found universally on M-HDRs. Every M-HDR also has a large, infinitely rotating wheel for scrubbing and data entry. The Vestax and Roland machines have a simple jog wheel. The Akai, E-mu, and Fostex machines have complete jog/shuttle wheels like those found on professional video-editing decks (e.g., the Panasonic AG-7500 series).

Obviously, a jog/shuttle wheel is better than a single jog wheel because it provides much more control over the speed at which data is entered or sound is scrubbed. With the inner, jog wheel, you can carefully dial things in. With the outer, spring-loaded shuttle wheel, you can move in quick bursts or in even, sustained sections.

The Akai, E-mu, and Roland machines also sport 10-key pads, which are excellent data-entry tools. I sorely missed this feature when using the Vestax and Fostex machines.

THE REMOTE SIDE OF TOWN

MIDI Machine Control is a standard feature for all M-HDRs except the DR8,

EM ON M-HDR

The first commercial modular hard-disk recorder was the Roland DM-80, which was initially released in 1990. Since then, ten more such units have been released. All but the Roland DM-80 are still in production.

Of these eleven M-HDRs, EM has reviewed all but the Akai DR16, which is identical to the DR8 except for having sixteen tracks; the Vestax HDR-8, which is an expanded version of the 6-track HDR-6; and the Fostex D-80, which was released just in time for this article.

Product	Issue
Akai DR4d	February 1994
Akai DR8	January 1996
E-mu Darwin	May 1996
Fostex DMT-8	April 1996
Roland DM-80	January 1993
Roland DM-800	October 1995
Roland VS-880	September 1996
Vestax HDR-6	August 1995

which requires the IB-803M MIDI expansion card. Through MMC, any of a number of remote control devices can be used, including a computer, Alesis BRC, E-mu Launch Pad, or JL.Cooper CuePoint.

A few manufacturers have control surfaces designed for their M-HDRs. Akai's DL16 remote control ($1,695) provides all the front-panel functions of the DR8 but improves the user interface by adding sixteen faders and pan pots, EQ controls, and a few dedicated edit buttons. Vestax's MX-1 ($499) has transport controls and a mixer section. Both devices connect via MIDI and could conceivably be used to control other MIDI devices through MMC.

The Fostex D-80 has a built-in remote-control surface, which is a very cool idea. Its entire front panel actually pulls off, remaining linked to the main unit via a multipin cable. The stock cable is too short to be useful (about six inches), but a 16-foot cable

can be purchased from Fostex for about $50.

The Akai DR8 and the Roland DM-800 can use an ASCII (computer) keyboard as a remote control, which is a neat feature. This comes stock on the DM-800; on the DR8 it is a feature of the Superview expansion card. All transport controls, locate points, and editing functions can be accessed from the keyboard.

MIXING IT UP

When equipped with an onboard mixer and effects, the M-HDR provides most of a recording studio in one box. Every M-HDR except the two units from Fostex has an onboard digital mixer, and the Fostex DMT-8 has an analog mixer. The mixers vary widely in features, especially in the number of inputs and whether equalization and effects are included. With one of these M-HDRs and a DAT machine, you can keep everything in the digital domain, from tracking to mastering. That's the idea, anyhow.

Some units do a better job of fulfilling this goal than others. Although the Darwin has a digital mixer for its internal tracks, the mixer does not accept additional external signals at mixdown, and it doesn't have EQ or

effects. The DR8's mixer has the most input channels: eight for the internal tracks and eight for external signals. (The same 16-channel mixer appears on Akai's DR16, allowing that machine to mix all sixteen internal tracks.) When equipped with an expansion board, the DR8 can have three bands of real-time, parametric EQ on each of its sixteen channels.

The HDR-8's mixer includes two independent equalizers on each track. Each EQ can be fully parametric or high- or low-shelving with adjustable frequency. The mixer has only four analog inputs, but they can be used simultaneously with its S/PDIF input, so the unit can record up to six signals at once. (S/PDIF is a 2-channel format.) The HDR-8 also ships with free Windows software that allows complete real-time automation of its internal mixer. No Mac version of the stand-alone software is planned, but mixer maps are available for Emagic's *Logic*, Opcode's *Vision*, MOTU's *Performer*, and Steinberg's *Cubase* sequencer families for the Mac.

Of all the modular HDRs I tested, only the two Roland units and the DMT-8 have traditional mixing surfaces with faders. The DM-800's mixing surface is pretty rudimentary, though. Its

MAGNIFICENT SPECS

	Akai DR8	E-mu Darwin	Fostex D-80	Fostex DMT-8	Roland DM-800	Roland VS-880	Vestax HDR-8
ADC	18-bit, 64x oversampling	18-bit, 128x oversampling	18-bit, 64x oversampling	18-bit, 64x oversampling	18-bit, 128x oversampling	18-bit, 256x oversampling	18-bit, 128x oversampling
DAC	20-bit, 8x oversampling	16-bit, 64x oversampling	18-bit, 64x oversampling	18-bit, 64x oversampling	18-bit, 8x oversampling	18-bit, 8x oversampling	20-bit linear
Internal Processing	24-bit linear	24-bit mixer, 32-bit DSP	n/a (no mixer)	n/a (analog mixer)	24-bit	24-bit	24-bit
Sample Rates (kHz)	32, 44.056, 44.1, 48	44.1, 48	44.1	44.1	32, 44.056, 44.1, 48	32, 44.1, 48	32, 44.1, 48
Frequency Response	20 Hz–22 kHz (+0/-1 dB)	20 Hz–20 kHz (±1 dB)	20 Hz–20 kHz (±1 dB)	20 Hz–20 kHz (±1 dB)	20 Hz–20 kHz (±1 dB)	20 Hz–20 kHz (±1 dB)	20 Hz–22 kHz (±1 dB)
THD (@ 1 kHz)	0.05% (full scale)	0.02% (A-weighted)	0.01% (full scale)	0.01% (full scale)	0.02% (@ -10 dBm)	0.07% (@ -10 dBm, in MT1 mode)	0.009% (@ -10 dBm)
SCSI	yes	yes	no	no	yes	yes	yes
Max. Addressable HD	4 GB per file	2 GB	1.7 GB	1.7 GB	8 GB	1 GB internal, 1 GB external	4 GB (in 2 GB partitions)
Audio Compression	no	no	no	no	no	yes (8 tracks with, 4 tracks without)	no

ALESIS

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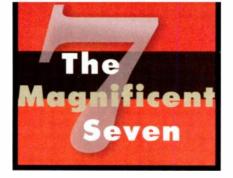
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ALESIS

mixer has real faders for the channels, stereo aux returns, and master L/R bus and uses knobs for panning and for the 2-band semiparametric channel EQ. (With a semiparametric design, the level and frequency can be changed, but the bandwidth can't.) The VS-880 sports a 14-channel mixer with a much more complete set of physical controls than the DM-800 has; faders and knobs abound. It has three bands of parametric EQ in Tracking mode or two bands in Mix mode.

The DMT-8 is another story entirely. The machine's subtitle is "Digital Multitracker," which indicates that Fostex conceived the unit in the tradition of the company's Multitracker-series cassette recorders. Figuratively speaking, you might call the DMT-8 mixer a chip off the old circuit board. In contrast to the digital mixers in the other HDRs, it offers a traditional 4-bus (8 × 4 × 2) analog mixer with eight direct outputs and eight unbalanced inputs, all on ¼-inch jacks. The unit has mic preamps and trim pots but no balanced inputs. Each channel has two aux sends and two bands of semiparametric EQ. Four channels have TRS insert points for use with external processors. It's not a pro mixer, but it's quite serviceable for the average home recordist.

All of the onboard mixers except those in the DMT-8 and Darwin respond to MIDI continuous controllers. The Darwin's mixer will gain this capability in the aforementioned November upgrade. Depending upon the machine, Control Changes can modulate everything from levels and panning to EQ and delay time. With one of these units and a sequencer, you can automate your mixes and keep them in the digital domain, a combination that has never been available in one piece of equipment in this price range.

The Roland and Akai machines also have various degrees of built-in automation. Roland's DM-800 has full real-time automation, programmable from its faders and knobs. The VS-880 only has snapshot automation, which controls levels, EQ settings, and effects settings. Akai's DR8 has an improved version of snapshot automation in which the track levels can be crossfaded between two snapshots. (A snapshot is a "picture" of all parameter values at a given instant.) This results in a type of continuous automation, but with a quirky interface and less fine control than conventional continuous real-time automation provides.

TIME AND SPACE

In addition to having EQ, the VS-880 includes a time compression/expansion algorithm that can be applied with or without pitch change. The DM-800 has time compression/expansion and a pitch-change algorithm that allows you to change the pitch of a selection without changing its duration. This powerful tool allows for pitch-correcting vocals without affecting their timing. The Darwin can do time compression/expansion and pitch change with the addition of the Model 4014 DSP expansion card.

Currently, you can add an effects-processor expansion card only to the Roland VS-880 and Vestax HDR-8, and E-mu has promised to release an effects expansion card for Darwin sometime in November.

The optional effects board for the VS-880 is exceptional and highly recommended. Its effects—mostly drawn from Roland's GP-100, SRV-330, and SDE-330 effects processors—include reverbs, delays, amp simulators, and such preconfigured multi-effects as chorus, flange, pitch shift, EQ, compression, 3-D soundstage enhancement, and de-essing.

Unfortunately, I wasn't able to test the Vestax HDR-8 digital effects–processor board. According to Vestax, it includes 128 patches and delivers reverb, chorus, delay, flanging, and several preset multi-effects. There are not a lot of editable parameters, but this probably reflects the board's very reasonable list price (see the table "Expanded Armory").

DODGING BULLETS

Of course, all modular HDRs have editing features and an undo function. These tools are among their principle advantages over digital tape recorders, allowing you to overcome goofs and fine-tune your work to an extent no tape recorder can. How comprehensive these features are varies widely from unit to unit. Keep in mind that using undos extensively can eat up a lot of disk space.

All but the Fostex machines have true nondestructive editing; that is, pointers to the data are changed, but the original data is not touched. Fostex's claims of having nondestructive editing are based on having an undo feature, but these machines actually write new audio files to the disk during editing.

With its powerful mixer, excellent MIDI implementation, flexible EQ, top-flight effects option, and well-implemented virtual tracks, the Roland VS-880 is a great choice for songwriters. The main downside: it lacks individual track outs and external expansion ports and can only sync to MTC.

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As a result, edits use more hard-disk space, and longer edits take a long time to process.

Every machine has cut, copy, and paste commands. The Vestax HDR-8 has a Merge function that digitally combines tracks faster than real time, providing an expedient way of bouncing tracks down. The DR8 and DM-800 have Slip functions for nudging selections in small increments of time. (In the DM-800, this is part of the Move function.) This is great for scooting a drummer who's playing ahead of the beat into the pocket.

The VS-880 and DM-800 have the most powerful alignment features. Using their To and From edit points, you can align any selection's start point, end point, or any point in its middle to any marker in a project. This is an invaluable tool when editing sound to picture (e.g., locking a car-crashing sound effect to a car crash on the screen). The DR8 has similar features, but they require the Superview monitor option.

When you edit nonmusical material, cutting to SMPTE times works fine. However, when you are editing music —say, copying a chorus to a different part in your song—it is really great to be able to make your cuts and splices on the beat (that is, to a tempo map). Roland and Akai machines can save tempo maps as part of a project's parameters. This allows you to switch the main time display from SMPTE to bars, beats, and clicks and perform your editing in a much more musical fashion. Although the Fostex machines don't sync directly to SMPTE, they do let you create a tempo map and edit using bars, beats, and clicks.

Some of these machines are easier to work on because of their superior user interfaces. The E-mu and Roland HDRs can present graphic representations of tracks in the form of blocks on their LCD screens. You can see which tracks have data on them and view the area where you want to perform an edit. The Roland machines also have an amplitude-waveform view-

ing feature that allows you to dial in zero-crossings, which is an excellent tool for creating silent in/out edit points.

Akai's Superview expansion card allows the DR8 to display visual representations of the waveforms on each track using an external monitor. The monitor also displays all the meters and nameable objects.

One of the best features of HDRs is their ability to undo a take; you can always reverse your most recent edit. It's sort of like musical bungee jumping: no matter which bridge you jump off, you know you can always bounce back. Most units have at least one level of undo; only the Roland DM-800 lacks this feature. On the other hand, the Roland VS-880 has an incredible 999 levels of undo and wins first place for "undo-ability." The Darwin comes in second place with up to sixteen levels of undo—quite sufficient for most purposes.

BEYOND THE BEATEN TRACK

Most hard-disk recorders let you attach several takes, or virtual tracks, to a single track. Even though you can only hear eight tracks at once with the machines I compared, you can have a slew of different versions hidden beneath the track you're listening to. For instance, if you're auditioning six vocalists but you only have one track open for all six to sing on, it's not a problem. Simply record each vocalist on a different virtual track; later, you can recall the different takes one at a time until you find the one you like. Imagine the possibilities: different solos on different remixes, different languages on the same commercial voice-over, different sound effects for the same picture. And all this is possible without erasing the original take. That's why virtual tracks are cool!

The Akai, E-mu, and Roland machines all have some sort of virtual track feature; the Vestax HDR-8 and

EXPANDED ARMORY

AKAI DR8 EXPANSION BOARDS

IB-801S	SCSI-B interface	$299
IB-802T	SMPTE interface	$379
IB-803M	MIDI interface	$299
IB-804A	ADAT Lightpipe interface	$299
IB-805R	RS-422 interface	$299
IB-806B	Biphase interface	$299
IB-807V	Superview	$699
IB-EQ16	16-channel EQ	$699
IB-EQ8	8-channel EQ	$550

E-MU DARWIN EXPANSION BOARDS

Model 4010	4-input expander	$249
Model 4011	ADAT digital I/O interface	$149
Model 4012	ADAT synchronization port	$379
Model 4014	DSP board	$495

ROLAND DM-800 EXPANSION OPTIONS

DA-400	output expander	$645
DIF-800	ADAT, DA-88, and RS422 interfaces	$1,095

ROLAND VS-880 EXPANSION BOARDS

VS8	multi-effects processor	$395

VESTAX HDR-8 EXPANSION BOARDS

AD-2	analog 2-input expander	$190
FX-1	multi-effects processor	$299
SE-1	SMPTE interface	$370

The Magnificent Seven

Fostex units do not. The DR8 is the least flexible of the units that have this feature, offering only five additional tracks (Akai calls them Takes) per project. Both Roland machines use the layer method, in which several tracks, or Takes, can be hidden directly beneath the active track. You can hear a different track simply by calling it to the surface by name.

The Darwin has what E-mu labels "virtual slave reels" (VSR). Instead of having tracks layered beneath tracks, entirely new versions (or copies) of the eight original tracks are cloned. Imagine making a copy of the master 8-track reel onto a different reel of tape and then slaving it with the original, and you have the idea. (Just remember that you only have an 8-track mixer.) These virtual tracks exist in the same time domain that the original does, so takes can be copied between the different versions simply by using the copy, cut, and paste commands.

The VSR concept appears to be the best approach to virtual tracks, organizationally speaking, but the layer method is the most intuitive to use. Grabbing a layer and calling it to the surface is really easy. However, none of the layers can be individually named, so remembering which layer includes a particular take gets really confusing when you've recorded a lot of layers. In the Darwin, each VSR can be named, so this problem doesn't arise.

IDIOSYNCRASIES

There are several levels and types of synchronization questions that must be addressed when shopping for an M-HDR. Does the machine support MTC, and if so, which frame rates? Can it function as both master and slave? Does the M-HDR support SMPTE? If so, will it generate and lock to direct time code? Can the machine lock directly to other popular digital multitracks (for example, the Alesis ADAT or TASCAM DA-88)?

All of the machines support MTC. They all work great as masters with just about anything that syncs to MTC, such as a drum machine, a keyboard workstation's sequencer, or a software sequencer. All of the M-HDR manufacturers except Vestax claim that their machines slave to MTC, as well. Be dubious of this claim; these machines do indeed slave to MTC, but the accuracy of the sync leaves something to be desired.

I'm not going to single out any particular unit because it seems that different software, hardware, and M-HDR combinations determine how tight the lock is, and I was unable to try every combination. I did find that Roland's machines sync well with other Roland products (e.g., the sequencers on the XP-series workstations) and that the Darwin liked MTC coming from Opcode's *Vision* 3.0 better than from Mark of the Unicorn's *Performer* 5.0. Don't ask me why.

Similarly, even though many manufacturers claim that their units slave to SMPTE, don't assume you can accomplish it through a SMPTE-to-MTC conversion. If all you see in the back of the HDR is an array of MIDI jacks and there isn't a SMPTE expansion card available for the unit, don't count on getting a subframe-accurate lock without some serious work. If you need a professional lock that you can always count on to be subframe accurate, don't settle for anything less than actual SMPTE inputs on the HDR.

The only units that currently connect directly with SMPTE are the HDR-8, DR8, and DM-800. Around the time of its November firmware upgrade, E-mu also expects to ship the model 4013 SMPTE sync card (price tba), which will give the Darwin this capability. The DM-800 is the only HDR that comes stock with a SMPTE reader/generator. In addition, the DM-800 resolves to the frame-edge of the time code rather than simply triggering at a given SMPTE address, which is especially desirable for post production. The DR8 can read and generate SMPTE with the addition of the IB-802T SMPTE interface expansion card. The HDR-8 expansion card reads SMPTE time code, but you'll need another device to generate it.

The ability to lock directly to ADAT or DA-88 can be an invaluable asset. Your creative freedom increases when you can keep your MDM tape recorders in the main studio and have a satellite M-HDR to take home or to gigs. Composing and performing outside the pressures of a pay-by-the-hour studio with the knowledge that whatever you do will sync back up to the main multitracks is the next best thing to sliced bread.

So far, only E-mu and Roland have managed to implement a proprietary lock with another digital multitrack. With the Model 4012 ADAT synchronization expansion card, the Darwin is able to lock directly to one or more Alesis ADATs. The E-mu machine can function as the master or a slave, and the lock between Darwin and ADAT is tighter than between two ADATs.

Roland's DIF-800, mentioned earlier, gives the DM-800 direct synchronization with TASCAM DA-88s and DA-38s and Alesis ADATs. I didn't get

The Vestax HDR-8 provides outstanding value, thanks to its expandability, capable mixer, dual-format S/PDIF I/O, effects option, SMPTE sync, and solid construction. However, it does not support virtual tracks.

GUTS AND GLORIES

	Akai DR8	E-mu Darwin	Fostex D-80	Fostex DMT-8	Roland DM-800	Roland VS-880	Vestax HDR-8
Price	$3,495	$3,195	$2,195	$1,995	$6,295	$2,495	$2,495
Tracks	8	8	8	8	8	8	8
Virtual Tracks	5 takes	800	0	0	600	64	0
Levels of Undo	1	16	1	1	0	999	1
Locate Points	9 + 100 Stack + 2 Last	40	8	8	40 markers	32 + 999 markers	8
Analog Inputs	8	4	8	4	4	4	4
Analog Outputs	8 direct, 2 aux sends, mix L/R	8 direct	8 direct	mix L/R, 2 aux sends	4 (can be direct; mix L/R, or aux send)	mix L/R, 2 aux sends	mix L/R, 2 aux sends
Digital I/O [1]	S/PDIF, AES/EBU, ADAT	S/PDIF, AES/EBU (RCA), ADAT	S/PDIF (optical)	S/PDIF (optical; backup only)	S/PDIF (1 in, 2 out)	S/PDIF	S/PDIF (RCA & optical)
Backup Options	SCSI, Audio DAT, ADAT	SCSI	Audio DAT	Audio DAT	SCSI, Audio DAT	SCSI, Audio DAT	SCSI, Audio DAT
Internal Mixer Channels	16	8	n/a	8	12	14	12
Channel EQ	3-band parametric	n/a	n/a	2-band semi-parametric	2-band semi-parametric	3-band (2 semi-parametric; 1 parametric)	2-band parametric
MIDI	MMC, CC	MMC	MMC	MMC	MMC, CC	MMC, CC	MMC, CC
Synchronization	SMPTE, MTC, MIDI Clock w/Song Position, RS-422, Biphase	ADAT, MTC	MTC	MTC	SMPTE, ADAT, DA-88, MTC, RS-422 Song Position	MTC MIDI Clock w/Song Position	SMPTE, MTC
Time Code Rates and Types (fps)	24, 25, 29.97, 30, 29.97d, 30d [2]	24, 25, 30, 30d	24, 25, 30, 30d	24, 25, 30, 30d	24, 25, 29.97, 30, 29.97d	24, 25, 29.97, 30, 29.97d	24, 25, 30, 30d
Expansion Ports	5 external	5 external, 1 internal	0	0	0	1 internal (holds 2 boards)	2 external

1. S/PDIF ports are RCA and AES/EBU ports are XLR unless otherwise indicated.
2. *d* signifies drop frame.

to test the DIF-800, but based on past experience with Roland gear, I'm confident the sync is very good.

RECORDED IN TECHNICOLOR

All the machines I tested sound great. However, each has its own sonic signature. The two Roland machines were the most pleasing to my ears. Oddly, they were the machines that colored the sound the most, but it was a warm, round coloration with a nice, fat bottom end. Even Roland's VS-880, which employs audio compression, sounded great. I didn't like high levels of com-

pression, but a little bit seemed to create an almost analog type of sound.

Akai's DR8 seemed to have the most accurate sound. It sounded clinically digital; whatever I recorded is what I heard back. The sound of the Darwin is indistinguishable from that of an Alesis ADAT XT.

The Fostex D-80 and Vestax HDR-8 sound fine. I wasn't quite as impressed by them as I was by the other machines because they appear to have slightly less bottom end, but we're discussing very subtle differences here. The two decks sound very similar to Fostex's

DAT machines. The Fostex DMT-8 is a good, clean machine, though to my ear it has a bit less punch in the low end and a slightly higher noise floor than the D-80. That's to be expected considering the DMT-8's analog mixer.

HARD DECISIONS

By far, the most professional M-HDR is the Akai DR8. This machine has all the goodies a professional musician or engineer needs. Its expandability, full complement of inputs and outputs, and SMPTE capabilities make it the kind of machine that will always keep up

The Magnificent Seven

with your professional needs. Akai's experience in the pro-audio industry is obvious. The DR8's only fault is that it doesn't support direct synchronization with the ADAT and DA-88.

The Roland DM-800 is also a professional unit, and it is the only modular HDR that supports synchronization with both ADAT and DA-88. However, this great-sounding recorder comes with a hefty price tag: more than $7,000 with the DIF-800 option. As mentioned earlier, though, the stock DM-800 includes most of the options found in the DR8. The DM-800 still is more expensive if you include comparable DR8 options, but the prices are not as far apart as they appear at first glance.

The Darwin is much more affordable. This unit has all the makings of a top-notch professional HDR. It supports ADAT (but not DA-88) synchronization, and its great user interface, full complement of ins and outs, expandability, VSR feature, and onboard Jaz-drive option make it one of the best. Although its internal mixer leaves something to be desired, its DSP expansion card, due out by the time this article hits the newsstand, should spice it up. This unit may be the HDR of the future; if E-mu continues its development as promised, the Darwin is going to be hard to beat.

In the less-than-$3,000 category of HDRs, Roland's new VS-880 is a truly incredible piece of gear—a real wonder box. With its internal mixer, great sound, MIDI, equalization, effects, Jaz-drive option, and 64 well-implemented virtual tracks, this baby is a songwriter's dream. Yes, it lacks important professional features: individual outs, audio compression, and external expansion ports. In addition, it only offers synchronization via MTC. (It has one internal expansion port.) But it is, without a doubt, the best composition tool I have ever used.

Although the Vestax name is rarely associated with high-end professional equipment, the HDR-8 is an impressive unit, especially considering its price tag. Because of its expandability, ruggedness, and affordability, this HDR offers a great value to someone who doesn't have a lot of money but anticipates adding professional features (e.g., more inputs or SMPTE slave capabilities) in the future. Overall, this one gets the nod for best value in terms of bang for the buck.

Fostex took a different tack with the D-80. It's the only modular HDR that has no mixer, allowing you to treat it like a traditional tape deck with a full set of track inputs and outputs. Because you aren't paying for a mixer, the D-80 is significantly less expensive than most of its competitors. Of course, it isn't a portable studio-in-a-box in the same sense as the other M-HDRs. In addition, its sync options are limited to MMC and MTC. On the other hand, the D-80 offers two very interesting, not-so-traditional features: a removable remote-control surface and a removable cartridge-encased hard drive. Rather than following the same path as everyone else, Fostex has given us a real choice.

Fostex also marched to its own beat when designing the DMT-8. Fostex's firstborn M-HDR stands in a unique position in that it is the only M-HDR exclusively aimed at the cassette-oriented home recordist. Its clean, friendly analog mixer and familiar user interface are its strongest selling points. It lacks many features that are important for the pro, including SCSI, removable media, mixer automation, and expansion ports. But the DMT-8 sounds very good, offers a full jog/shuttle wheel, and is the least expensive modular HDR.

So there you have it: the true story of the Magnificent Seven. (Cue *The Magnificent Seven* theme, and roll the credits!) Each modular HDR has its advantages and drawbacks; each is the right machine for someone. If you're serious about digital recording, the chances are excellent that one of them is right for you.

Erik Hawkins *would like to thank Michael Denten for his assistance with this review. Come visit us at the new Infinite Studios in Alameda, California, and look for us on the Web in December.*

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Yamaha's XG sound modules offer a wealth of musical possibilities.

Multitimbral muse

By Clark Salisbury
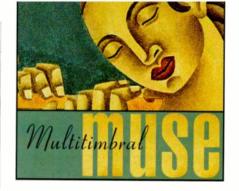
If you've checked out the reviews of Yamaha's new XG sound modules, the MU50 and MU80 (the latter was reviewed in the December 1995 **EM**), you know that I'm not the only one who thinks these little devils sound great. With General MIDI (GM) compatibility, crystal-clear sound, low noise, kickin' drums, and a flexible effe   system, what more could you want?

Well, programmability would be nice. To be fair, the MU series provides a long list of parameters that can b   edited from the front panel or via MIDI Control Change (CC) or System Exclusive (SysEx) messages. In fact, there are many hidden parameters that can't be accessed from the front panel at all. However, they can be tweaked mercilessly via MIDI if you know the code.

Even so, the MU series provides no user RAM to store edited sounds. So what's a tweakologist to do? If you own a sequencer that records and plays CC and SysEx messages, you can use ˙ to store and perform parameter changes. You can set up these parameter changes at the beginning of a sequence, which automatically (and rapidly) reprograms the MU at the touch of the play button. You can also use the sequencer to vary parameters in real time during the course of a performance. And of course, you can synchronize these changes to your sequence.

Illustration by Jeanne Berg

FIRST THINGS FIRST

The information presented here is intended to help you get the most out of your XG tone generator. At this writing, these devices include the MU50 and MU80 MIDI modules; QS300, CS1x, and CBX-K1XG keyboard synths; DB50XG daughterboard for the Sound Blaster and compatible sound cards; SW60XG synth card for PC-compatible computers; and the new QY700 sequencer/tone generator. The CC and SysEx messages provided here work with all of these devices, but references to specific buttons on the front panel apply only to the MU50 and MU80.

Let's start with a brief definition of XG, which is Yamaha's enhancement of GM. (XG instruments are fully GM compatible.) To conform to the XG standard, a sound module must be capable of at least 32-voice polyphony (the MU80 is 64-voice polyphonic) and 16-part multitimbral operation. (The MU80 and QY700 offer 32-part multitimbral operation with two MIDI ports, which is fully supported in the XG spec.)

The minimum XG sound set includes 480 patches and eleven drum kits, requiring the use of Bank Select (CC 0 and 32). In addition, XG instruments can utilize up to four drum kits on different channels simultaneously, and each kit is independently editable.

All XG modules also include at least three effects processors: reverb, chorus, and "variation," which can include a wide variety of miscellaneous effects. The variation processor can apply to the entire system or to a single part as an insert effect. The MU80 includes a fourth, insert processor that provides distortion effects for any single part.

Finally, most XG instruments also include a stereo analog audio input that is mixed with the internal sounds at the module's output. The MU80 and SW60XG go one step further by digitizing the signal and passing it through their internal effects. The A/D input on these two devices responds to Control Change and Program Change messages to call up the desired effects and to control volume, panning, and other parameters for the external signal. An audio input is not required to comply with the most basic implementation of XG (called Level 1). For example, the QY700 and QS300 have no audio input.

Some parameters fall under the general heading of "setup": pan positions, effects levels, initial volumes, and so on. For instance, when starting a new sequence using an XG module, I recommend resetting the instrument. Doing this guarantees that you'll start with a clean slate, so that some leftover parameter values from that experimental industrial schottische you were working on last week won't come back to bite you on the decrement button.

If you intend to play your music on other GM modules, send both the standard GM On message (F0 7E 7F 09 01 F7) and the XG On message (F0 43 10 4C 00 00 7E 00 F7) to the module. (These are SysEx messages in hexadecimal form; most modern sequencers let you enter any SysEx string and send that string to a MIDI device. If you need some help understanding hexadecimal notation, see "From The Top: SysEx Hex" in the January 1994 **EM**.) These messages reset the entire instrument to the default factory condition. Any GM devices that do not conform to XG will simply ignore the XG On message. If you're never going to play your sequence on anything but an XG module, you can omit the GM On message.

Figure 1 illustrates a typical setup sequence that starts with these SysEx commands, which apply to the entire device and should therefore be sent only once. The remaining messages should be sent to the module on all sixteen MIDI channels. I keep a copy of this sequence as a starting template for any XG music I create.

The channel messages start with two Bank Select messages (the Most Significant Byte, or MSB, and the Least Significant Byte, or LSB) and the desired Program Change. Following these are the initial Volume and Pan settings (CC 7 and 10, respectively). I recommend setting the overall track volume with MIDI Volume and then using CC 11 (Expression) to dynamically control

FIG. 2: In this example, NRPNs are used to adjust vibrato rate.
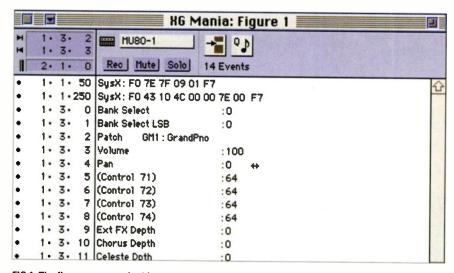
FIG 1: The first two events in this setup sequence are the SysEx GM On and XG On commands. Note the timing of these events as shown in the left column; plenty of time has been left after each message to let the receiving module go through its setup routines.

the track's volume during the sequence. In **Figure 1**, the Volume is set at 100 to leave some headroom.

The next four Control Changes—71, 72, 73, and 74—are used to control filter resonance, amplitude envelope release time, amp envelope attack time, and filter cutoff, respectively. The value for each of these controllers is set to 64, which maintains the values programmed with each patch; increasing the value from 64 offsets the preprogrammed value by a positive amount, and decreasing the value offsets by a negative amount. For example, to raise the filter cutoff, send a CC 74 value greater than 64; to lower the filter cutoff, send a value less than 64. To reset the filter cutoff to its original preprogrammed amount, send a value of 64.

The last three messages (CC 91, 93, and 94, which are also known as Ext FX Depth, Chorus Depth, and Celeste Depth) set the effects level for the reverb, chorus, and variation effect. In addition, the MU80 contains one extra effect (distortion) and a multiband EQ that can be adjusted via SysEx commands (more on this later). The distortion and multiband EQ effects are not present in the MU50.

TWEAKING A SOUND

Making changes to XG sounds using this template is as easy as getting a politician to mention his own name. If you want to shorten the attack of a string sound, just decrease the value of CC 73. Want that synth bass sound to break wind like a mutant duck? Increase the value of CC 71 (which controls the Resonance parameter). Here's a helpful hint: if you're having trouble bringing a sound forward in a mix, try increasing its resonance a little. Don't overdo it, though; too much resonance can be quite annoying.

Although XG modules support a slew of MIDI Control Changes (check the owner's manual for a complete list), some parameters do not have a dedicated CC to control them. However, many parameters can be changed using special CC messages that are called Registered Parameter Numbers (RPNs) and Non-Registered Parameter Numbers (NPRNs).

RPNs and NRPNs were introduced to increase the number of MIDI controllers without having to venture outside the current MIDI spec. The only difference between RPNs and NRPNs is
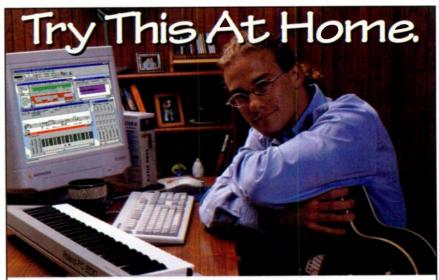
The MU80 64-voice tone module introduced Yamaha's XG standard, which is a superset of General MIDI. Most of the module's parameters can be programmed from the front panel, but some can only be accessed via SysEx.

that each RPN has a standardized function whereas the function of each NRPN is determined by the manufacturer. In both cases, a message is sent to select a parameter, and then CC 6 (Data Entry) is used to change the value of the selected parameter.

The message that selects a parameter includes two parts: the MSB and the LSB. For NRPNs, the MSB is specified by CC 99, and the LSB is specified by CC 98; for RPNs, the MSB is specified by CC 101, and the LSB is specified by CC 100. To select a particular parameter, you must send the NRPN or RPN MSB and LSB that correspond to that parameter.

Fortunately, this task is easier to do than it is to think about. For example, change the vibrato rate using NRPNs. Start by looking up the MSB and LSB of the parameter you want to address in

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the table "Handy NRPN/RPN Chart," and plug in the appropriate numbers. For vibrato rate, the NRPN MSB is 1 and the LSB is 8. Send the following messages on the channel to which the desired patch is assigned:

CC 99, value 1

CC 98, value 8

Pay attention to the order of these events; it's critical. These messages tell the XG module to change the vibrato rate for the patch in response to subsequent CC 6 (Data Entry) messages.

If this isn't perfectly clear, take a look at **Figure 2**. This short sequence sends the vibrato rate MSB (CC 99, value 1), followed by the LSB (CC 98, value 8), followed by a Data Entry message that actually affects the selected parameter (CC 6, value 72). After you send this

NOW YOU TELL ME

As much fun as it is to type long, seemingly meaningless strings of SysEx into your sequencer while trying to remember arcane NRPNs, there are a couple of programming aids available for XG modules. If you use a Mac and Emagic *Logic* or MOTU *Performer,* there are templates available to ease the pain of data entry. Both templates were developed by the inimitable Nate Tschetter, and they provide access to most of the parameters any sane person might want to control. Contact Emagic to inquire about the *Logic* template or Mark of the Unicorn if you're interested in the *Performer* template.

If your computer is of the PC ilk, don't despair; help is at hand in the form of *XGEdit,* written by Gary Gregson and available online for downloading from his Web page (http://www.cybertheque.fr/galerie/GGregson).

stream of messages, the rate of the LFO increases. (Remember, Data Entry values higher than 64 increase the parameter from the preprogrammed value, and values lower than 64 represent a negative offset. A value of 64 resets the parameter to its preprogrammed value.)

Once you've sent the MSB and LSB to select the parameter to be affected by Data Entry messages for a particular MIDI channel, the module will contin-

ue to respond to CC 6 messages in this way until you tell it not to. This can be a good thing. For example, if you're sequencing an electric-guitar track, you might find it handy to reset the sound's pitch-bend range during the sequence simply by sending new values for CC 6.

However, this can be a bad thing, too. For example, some instruments send CC 6 messages whenever the data slider is moved, which could send your guitar track out of tune every time you

adjust any parameter on your master keyboard. Sooner or later, you'll want to tell the module to stop responding to CC 6. Fortunately, RPNs and NRPNs can be "nulled out" by sending an MSB with a value of 127, followed by an LSB with a value of 127. To null out an RPN, send this string of messages on the appropriate MIDI channel:

CC 101, value 127
CC 100, value 127

To null out an NRPN, send this string on the appropriate MIDI channel:

CC 99, value 127
CC 98, value 127

DRUMS IN THE DISTANCE

NRPNs are really cool when used with XG drum kits because they let you control the individual drum sounds. In this application, the MSB specifies the parameter and the LSB specifies the note number affected by that parameter. (See the table "Handy NRPN/RPN Chart" for a list of drum-specific NRPN values.) For example, to turn the reverb off for the kick drum assigned to note number 36, send this string of messages on MIDI channel 10:

CC 99, value 29
CC 98, value 36
CC 6, value 0

Interestingly, individual drum reverb and chorus send levels do not respond to Data Entry messages in the same way that most parameters in XG modules do. Instead, the send levels correspond directly with the Data Entry values. For example, if you send a Data Entry value of 0, the send level will be 0; if you send a value of 64, the send level will be 64, not its preprogrammed value.

To tune the snare drum found on note number 38 down by two half-steps, send this string:

CC 99, value 24
CC 98, value 38
CC 6, value 62

In this case, the Data Entry (CC 6) value offsets the preprogrammed parameter value from 64, so a Data Entry value of 62 lowers the Coarse Tuning parameter by two half steps.
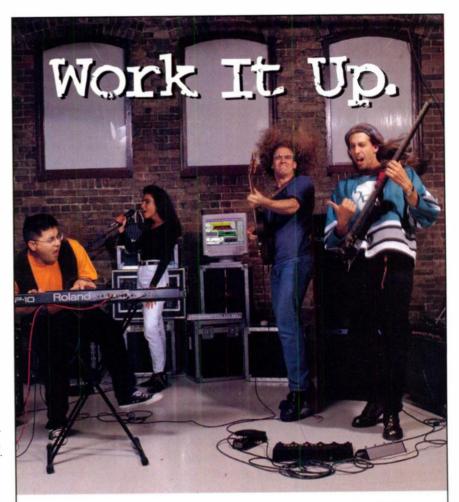
To darken the timbre of that snare drum, lower the filter cutoff by sending this string:

CC 99, value 20
CC 98, value 38
CC 6, value 42

Again, the Data Entry value offsets the preprogrammed parameter value.

Of course, you can do all this from

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the MU50/80's front panel. But there are applications that wouldn't be so easy without using MIDI controllers. For example, you can make an authentic-sounding train-type pattern (the kind used in lots of country and blues music) by specifying a relatively long EG attack time to soften the snare drum on upbeats.

Figure 3 shows two views of the same measure of sixteenth notes played on the snare. The graphic editing window shows the snare hits along with the CC 6 data in the strip chart across the bottom of the window. The event list gives a more complete (though somewhat more confusing) view of what's going on. An NRPN MSB with a value of 22 tells the module that any subsequent Data Entry messages are meant to control drum EG attack time, and an LSB with a value of 38 tells the module to

apply those messages to the drum assigned to note number 38. Then, the CC 6 value alternates between 16 and 48, which changes the snare's attack time during the measure.

NOTHING SUCCEEDS LIKE SYSEX

What about the XG parameters that don't respond to Control Change messages? As you might have guessed, the answer is SysEx. But don't worry. Using SysEx to control XG parameters is not as difficult as you might think, especially with the MU50/80. If you're using one of these modules, just select the parameter you want to change from the front panel, adjust it to your liking, and double-click the Enter button. This displays the SysEx message that will adjust the selected parameter to the specified value. Now, type this message into your sequence. When you play the sequence, the message will be sent, changing the parameter to the desired value.

For example, let's say you want to adjust the decay time of the reverb. Press the Effect button, select REV, press Enter, and use the Select buttons to scroll to the Reverb Time parameter. If
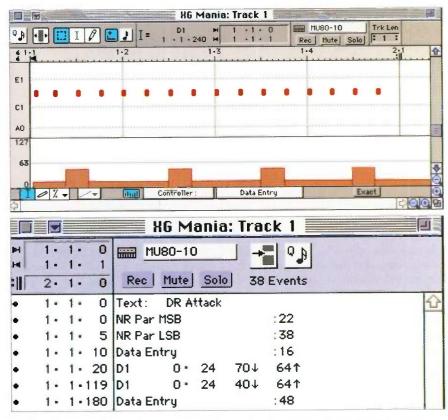
FIG. 3: In these two views of the same snare drum pattern, the graphic window (top) is easier to understand but not as complete as the event list (bottom).

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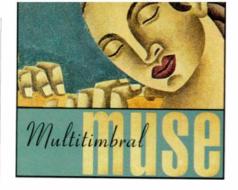
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you happen to be using the default reverb (Hall1), the parameter value will probably be 2.1. Adjust it to something else with the Value buttons, say, 3.0.

Now, double-click the Enter button, and the display shows the SysEx message (in hexadecimal) that will set the Reverb Time to this value. Simply enter the displayed values (in this case, F0 43 10 4C 02 01 02 1B F7) into your sequence in the appropriate place. When the module receives this message from your sequencer, it will immediately set Reverb Time to 3.0.

If you find yourself making lots of edits to the effects section of the MU50/80, you might find it a bit tedious typing all that SysEx into your sequencer. Instead, try this: after you tweak all of the effects parameters from the front panel to achieve the sound you want, send the following SysEx strings from the sequencer to the unit:

F0 43 20 4C 02 01 00 F7
  (requests all reverb parameters)
F0 43 20 4C 02 01 20 F7
  (requests all chorus parameters)
F0 43 20 4C 02 01 40 F7
  (requests all variation effect parameters)
F0 43 20 4C 03 00 00 49 F7
  (requests all distortion effect parameters, MU80 only)

When the module receives each of these strings, it sends a SysEx message that contains all of the corresponding parameter values back out to the sequencer. Enter the request messages into one track of a sequence, record enable another track, and start recording. When the module receives the SysEx messages from the first track, it dumps its effects parameters, which are recorded into the second track. You can then copy and paste these messages where you need them.

Now, when you play the sequence, these response messages will be sent to the module, causing the effects to be configured accordingly. Doing this is a lot easier than typing in all of the SysEx parameters one at a time. Just don't forget to mute the track with the original request messages after you have the MU50/80's responses. In addition, make sure there's plenty of space between these messages, perhaps as much as a quarter note. The module needs time to spew a pretty good-sized wad of data after each request.

STUPID SYSEX TRICKS

You can use SysEx messages to make the MU50/80 do many things besides upchuck effects parameters if you know the code. Note that you can't do any of these things from the front panel; only SysEx commands can subjugate the module to your every whim.

Turn the Multi EQ into a parametric (MU80 only). Normally, you don't have much control over the frequencies of the five bands in the MU80's master EQ; you can only select a basic EQ type (Jazz, Pop, Rock, etc.) and adjust the amount of boost or cut for each of the bands. If you know the SysEx, though, you can scoff at the preset EQ frequencies and establish entirely new EQ curves of your own. Check out the table "Handy Multi EQ SysEx Chart" to see how this works.

Now, I know some of you are saying, "How do I actually *apply* this stuff?" And I know there are others saying something like, "*Aaaaiiieeooowwweeee!!!!! Don't make me do SysEx!!!!!!*" But really, it's not that hard. These SysEx messages conform to the following format:

F0	SysEx status byte
43	Yamaha ID number
1n	Device Number
4C	Model ID (both MU50 and MU80)
aa	Address High
aa	Address Mid
aa	Address Low
dd	Data
.	.
.	.
.	.
dd	Data
F7	End of Exclusive

All you need to do is fill in the variables (1n, aa, and dd) with appropriate

Handy NRPN/RPN Chart

Patch NRPNs	MSB (CC 99)	LSB (CC 98)
Vibrato Rate	1	8
Vibrato Depth	1	9
Vibrato Delay	1	10
EG Decay Time	1	100
Null	127	127

Drum NRPNs	MSB (CC 99)	LSB (CC 98)
Filter Cutoff	20	*
Filter Resonance	21	*
Attack Time	22	*
Decay Time	23	*
Coarse Pitch	24	*
Fine Pitch	25	*
Level	26	*
Pan	28	*
Reverb Level	29	*
Chorus Level	30	*
Variation Level	31	*

RPNs	MSB (CC 101)	LSB (CC 100)
Pitch Bend Range	0	0
Fine Tune	0	1
Coarse Tune	0	2
Null	127	127

*LSB specifies the note number affected by subsequent Data Entry messages.

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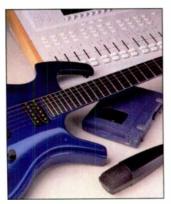
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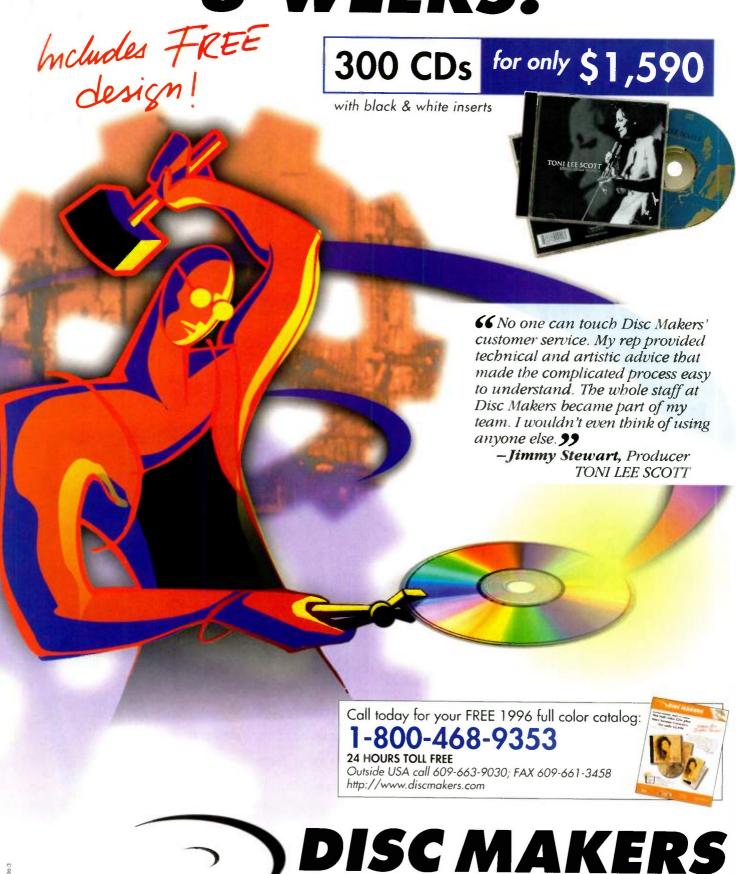
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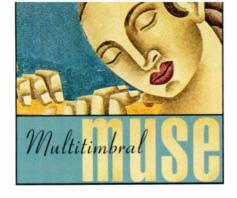
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Multitimbral **muse**

numbers (in hex, of course) and you're set. The first variable, 1n, refers to the device number. This number should be set to 10 unless you've changed the device number from the factory default for some reason.

The next set of variables is given in the chart. These variables specify the parameter you want to address. Note that the high and mid addresses remain the same (02H and 40H) for all Multi EQ parameters. The low address is used to select specific parameters in the Multi EQ.

The last variable, Data, depends on the nature of the parameter. The range

of acceptable values is found in the Value Range column of the chart, and a description of what this range actually represents is given in the Description column.

So how do you use this chart? For example, to set the frequency of the second EQ band (EQ frequency2, listed in the Parameter column of the chart) to its maximum value of 10.0 kHz, you would send the following message:

F0	SysEx status byte
43	Yamaha ID
10	Device Number
4C	Model ID
02	Address High
40	Address Mid
06	Address Low
36	Data
F7	End of Exclusive
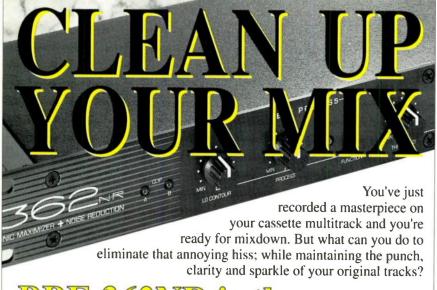
Apply random panning to a sound. Send the message F0 43 10 4C 08 nn 0E 00 F7 to cause the sound on the channel you're addressing to appear randomly in the stereo field with each new

note. In this case, nn specifies the MIDI channel you wish to address; 00 = channel 1, 01 = channel 2, and so on up to 0F = channel 16.

There's loads of other stuff you can get your XG module to do if you just know the SysEx. For example, you can use Channel Aftertouch to control volume by sending F0 43 10 4C 08 nn 4F dd F7, where nn is the MIDI channel of the sound you want to affect, and dd is the depth of control you want to exert from 00 to 7F. A similar message can be sent to assign Aftertouch to control the filter cutoff: F0 43 10 4C 08 nn 4E dd F7. To disable these assignments, just send the message again with dd = 40 (hex), which restores the default value.

You can also use Channel Aftertouch to control vibrato (PMOD) and tremolo (AMOD). To assign Aftertouch to control vibrato, send F0 43 10 4C 08 nn 50 dd F7; to assign Aftertouch to control tremolo, send F0 43 10 4C 08

Handy Multi EQ SysEx Chart

Address (hex)	Value Range (hex)	Parameter	Description (hex)	Default Value
02 40 00	00 to 04	EQ type	00: FLAT 01: JAZZ 02: POPS 03: ROCK 04: CLASSIC	00
02 40 01	34 to 4C	EQ gain1	-12 to +12 (dB)	40
02 40 02	04 to 28	EQ frequency1	32 to 2,000 (Hz)	0C
02 40 03	01 to 78	EQ Q1	0.1 to 12.0	07
02 40 04	00 to 01	EQ shape1	00: shelving 01: peaking	00
02 40 05	34 to 4C	EQ gain2	-12 to +12 (dB)	40
02 40 06	0E to 36	EQ frequency2	0.1 to 10.0 (kHz)	1C
02 40 07	01 to 78	EQ Q2	0.1 to 12.0	07
02 40 08	not used			
02 40 09	34 to 4C	EQ gain3	-12 to +12 (dB)	40
02 40 0A	0E to 36	EQ frequency3	0.1 to 10.0 (kHz)	22
02 40 0B	01 to 78	EQ Q3	0.1 to 12.0	07
02 40 0C	not used			
02 40 0D	34 to 4C	EQ gain4	-12 to +12 (dB)	40
02 40 0E	0E to 36	EQ frequency4	0.1 to 10.0 (kHz)	2E
02 40 0F	01 to 78	EQ Q4	0.1 to 12.0	07
02 40 10	not used			
02 40 11	34 to 4C	EQ gain5	-12 to +12 (dB)	40
02 40 12	1C to 3A	EQ frequency5	0.5 to 16.0 (kHz)	3C
02 40 13	01 to 78	EQ Q5	0.1 to 12.0	07
02 40 14	00 to 01	EQ shape5	00: shelving 01: peaking	00
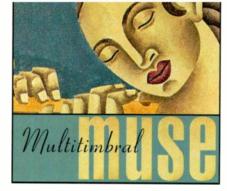
Multitimbral **muse**

nn 52 dd F7. As before, nn should be replaced with the channel you want to address, and dd should be replaced with the depth of control you want to exert, from 00 to 7F. A value of 00 disables the assignment.

All this stuff is listed in those charts at the back of the owner's manual that you never looked at carefully because it didn't make any sense. Hopefully, it will now.

MORE?

There's a lot more info on XG out there. If you have access to the Internet, you simply must visit the Unofficial Yamaha MU80 Home Page (http://arachnid.ccs.uwo.ca/mu80). Another great site is the MIDI Farm (http://www.midifarm.com), where you'll find all kinds of information on the MU modules plus links to many other cool MU/XG sites and software.

If you're in a more corporate mood, surf over to Yamaha's official XG site (http://www.yamaha.co.jp/english/xg/index.html). Also, Mike Overlin of Yamaha hosts a couple of XG forums: one on Compuserve in the MIDI A Vendor section and the other on the Microsoft Network (type Go Yamaha XG). In addition, Overlin has just started a new forum on AOL (keyword Yamaha XG).

For the technical-document *gourmand*, try the Yamaha Strategic Business Alliance Web site (http://www.ysba.com). You'll be required to join up, but there's no charge, and you'll gain access to a wealth of information (some of it rather technical) about XG products.

Finally, check out "Sizzling Sequences" in the February 1996 **EM** for several detailed examples of ways to use CCs, NRPNs, and RPNs in your sequences. Combined with the information presented here, you should have no trouble making beautiful music with your XG tone module.

Clark Salisbury *would like to thank Nate Tschetter for his help with this article.*

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06 | Reverse Reverb | L-Dly/ R-Gate Rev
07 | Chorus | L-Chorus/ R-Gate Rev
08 | Flange | L-Flange/ R-Gate Rev
09 | Delay | L-Rev & Slap/ R-Fla
10 | Dual Delay | L-Rev & Slap/ R-Chor
11 | Doubler | L-Reverb/ R-Flange
12 | Pitch Shift | L-Reverb/ R-Chorus
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Simply Grand

Practice these
techniques
for making
gorgeous
grand piano
samples.

Strictly speaking, the modern piano is most accurately defined as a percussion instrument: sound is produced when a hammer strikes a set of strings. The instrument came into existence in the eighteenth century as a "new and improved" harpsichord, when Italian inventor Bartolomeo Cristofori built what he called a *gravicembalo col piano e forte* (translated literally as "harpsichord with soft and loud"). Although the harpsichord used quills to pluck the strings, Cristofori ingeniously substituted hammers, in essence building a large-hammered dulcimer with a much greater dynamic range than the harpsichord.

Melodically, however, the earliest pianos were more limited than their modern descendants. We know that the piano owned by Mozart had just five octaves and that Liszt was playing an instrument with only six octaves as late as 1824. Of course, the modern piano has 88 notes, which encompasses a little more than seven octaves.

By Jim Miller

Illustration by Tana Powell

Simply Grand

This large pitch range has always made it especially difficult to sample pianos effectively. Fortunately, today's samplers have huge amounts of RAM available—for example, the Kurzweil K2500 and E-mu E4 can be loaded with up to 128 MB—which, theoretically, lets us do a much better job of accurately capturing the sound of a grand piano.

Of course, there are many commercial piano samples on the market. However, you might need a different type of sound, or perhaps you're just the DIY type. If so, it's time to create your own piano samples. All you need is the right information, which I will be happy to provide.

THE LIVE EXPERIENCE

Before we start, keep in mind that even the most exquisite piano sample cannot compare to the sound of playing the real thing. You can blame physics for this slap in the face of modern technology. When you sit down at a 9-foot grand and strike a note, you not only hear that note, you *feel* it. The vibrations of the strings and the resonance of the wood produce myriad sound waves that strike your body, and you can also directly feel vibrations as your fingers touch the keys. This experience cannot possibly be duplicated by a piano sample reproduced from a speaker or monitor system.

There are other factors involved, as well, such as the ratio of direct to reflected sounds within the piano's acoustic environment. If you're playing a 9-foot grand in a fair-sized room, the sound reflections are rather substantial and should envelope you in an ambient womb of harmonic delights. Once again, it is difficult to reproduce these psychoacoustic wonders with close-field monitors in a small studio.

So then, do you simply throw your hands in the air and give up? Not if you're one of the hardy breed of **EM** readers! Just because your samples will never approach the *physical* experience of playing a great piano doesn't mean you can't produce a meaningful *aural*

experience. By employing good miking and sampling techniques, you can emulate the beatific sound of the grand piano.

CHOOSING THE PIANO

There *is* one caveat to that previous statement, however: if you don't have access to a really fine grand piano— one that is superbly built, well maintained, and in near-perfect tune—you will never, *ever* create a great set of samples. You simply must start with the best instrument you can get your hands on.

So how do you choose the right piano to sample? First, you must decide exactly what type of sound you're after. Pianos have distinctive personalities, and pianists with good ears can easily distinguish between a Yamaha and a Steinway or between a Bösendorfer and a Baldwin.

For today's music, all sorts of different piano sounds are needed. If I were producing a country song in my studio, I would want the piano to punch through a mix of bright acoustic and electric guitars, so I might choose a bright Yamaha over a mellow Bösendorfer. For a new age session, however, the Bösendorfer would be perfect.

I recommend that you listen to a number of pianos before deciding which one you want to sample. If you live in a major metropolitan area, there may be several recording studios that have well-maintained pianos set up and ready to go. Just call around and ask which type of piano each studio owns and what they'll charge you per hour to come in and record it. If you can work without a staff engineer, the room rates for most studios are usually quite reasonable.

If you are lucky, you might even be able to use the studio's mics, and if you're *really* lucky, the mics may already be set up in position. But even if the studio has some classic Neumanns, bring along your own mics, and use them to record a second pass as a backup. You'll end up with an ad-

ditional recording that will be quite different texturally, particularly if you move your mics farther out into the room or closer to the strings.

You also might want to call your local university or music school and find out what they have available. In addition, many churches own beautiful pianos, and they might be willing to let you record them during the week in exchange for a small donation. If all else fails, check with your local piano retailer. If you walk in when the store is fairly empty, the staff will usually be gracious enough to let you set up and do some recordings. Just cross your fingers and hope that nobody decides to start vacuuming the floor while you're working.

In any event, choose the best piano you can find, or if you're fortunate enough to have access to several, go for broke and record all of them. Of course, *always* record to a permanent archival medium such as DAT, rather than directly into your sampler. Recording to DAT ensures that you'll have all the time in the world to select the best notes to use for your samples. In addition, if you add more RAM to your sampler at some later date, your library of piano sounds will come in quite handy.

FIG. 1: Close miking over the strings results in a bright, percussive sound.

Simply Grand

For example, if you currently have only enough memory for a monophonic piano sample, you'll be able to go back to your DAT source and create a dazzling stereo piano, once you increase your RAM.

MIC TECHNIQUES

As in most recording applications, high-quality mics often deliver a better finished product. The mics of choice for recording pianos in most studios seem to be any of Neumann's large-diaphragm condensers. (Hey, how can you go wrong with those?) However, I've had tremendous success with my beloved AKG C 414s, which I used to record the critically acclaimed Stereo Grand for the Alesis QuadraSynth Plus Piano. Recently, I used a Crown SASS-P (Stereo Ambient Sampling System PZM) for what has become my favorite piano sample.

If you're using two mics, check for phase cancellation before you start recording. Pan both mics to center on your mixer, and make sure that the lows or mids don't get lost. In addition, listen for a thin, nasal quality, which is another symptom of cancellation. Your best bet is to have someone play the piano while you adjust the mics to the positions that sound best through headphones. If you're on your own, record a section of music and then play it back to check the phase. You should also make sure the signals aren't clipping.
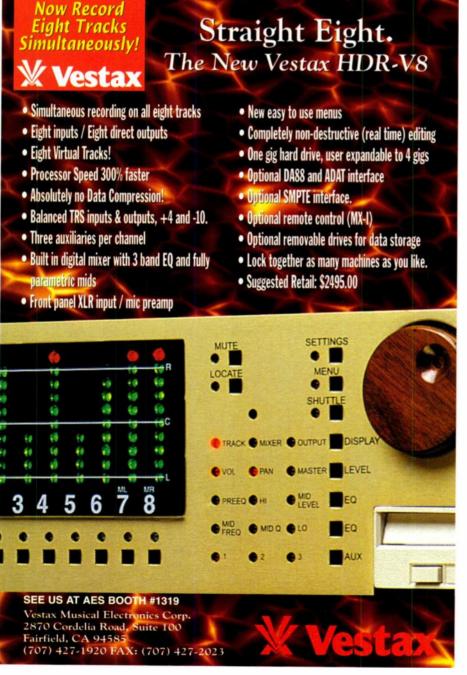
If you hear any sonic anomalies, change the position of the mics. If you hear distortion, move the mics farther away from the strings or activate a -10 dB pad (if your mic or console has one) to reduce transient levels, and then try again. Mic placement is the most critical part of the recording process. As a starting point, try using the

> **Better mics mean a better finished product.**

setup illustrated in **Figure 1**, and keep the mics about one to two feet from the strings.

It's important to use a mic's cardioid pattern (provided it's a multipattern mic) rather than the omnidirectional or hypercardioid modes. An omni-pattern mic may pick up too much of the signal that is intended for the adjacent mic, which can "blur" or "smear" the sound and cause phase problems. Hypercardioids, on the other hand, focus too strongly on the strings that are directly in front of the mic, so strings that are farther away from the mic's position can sound thin or weak and lack punch.

Of course, you don't *have* to close mike the piano. Another approach is to position your mics by the player's head (see **Fig. 2**). I've had tremendous success with this technique, particularly when using the Crown SASS-P, which is specially designed to create a

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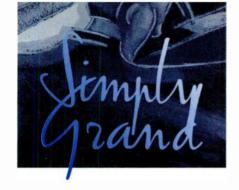
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strong stereo image. Although this mic position sacrifices a little in the "punchiness" department, it captures a lush, spacious re-creation of what the pianist actually hears during a performance.

If you have the resources, try both techniques separately, combine the two positions, or even move the "player position" mics to the "audience position" (see **Fig. 3**). The more perspectives you capture on tape, the more likely you are to end up with a great final product. A close-miking approach typically produces a bright, percussive sound that's perfect for rock, country, and fusion; room miking tends to work better for classical, jazz, and new age applications.

WHAT ABOUT DYNAMICS?

The grand piano can speak softly or quite loudly, and each dynamic level creates a very different set of overtones. When a key is played softly, the fundamental note predominates, whereas a loud performance produces a bright sound that is rich in upper harmonics.

A sampler's lowpass filters can certainly smooth out and warm up a bright sample to simulate a softer dynamic, but the end result will never be as realistic as a sample of a piano played softly. In addition, when you sample different notes for the soft and hard strikes, you add complexity to the final sound. For example, if you use an E♭ for the hard strike, use a D or E for the soft strike. The subtle differences may not be readily apparent, but they'll be there, adding just a bit more character to the final set of sounds.

Some sample collections include soft and hard strikes in addition to one at mezzoforte. If you have enough memory available, adding an "in between" dynamic can only help. Again, if that midlevel strike is a different note than the soft and hard strike, you add complexity. Velocity switching between a *very* soft strike and a *very* hard strike sounds much too extreme—too dark and too bright. It's better to bring your dynamic selections a bit closer together if you're going to use only two.

It is my personal feeling that more individually sampled notes, not more dynamic levels, produce the best overall sound. For example, in the custom grand piano sample I created for Constance Demby's latest album, *Aeterna,* I used only two strikes: mezzopiano and double forte. However, I sampled many different notes (in some places, every second or third note was sampled), and the finished product is shockingly realistic. Indeed, several reviewers stated that the album contains sampled sounds along with a real piano. (It doesn't.) I strongly believe this perception is due to the sheer number of sampled notes. In addition, the piano I sampled is an incredible instrument, which also helps.

NOTE FOR NOTE

In addition to considering dynamics, give some thought to the specific notes you will ultimately use in the final product. Simply sampling every third or fourth semitone does not typically produce the best results. Listen carefully and sample the notes that sound the sweetest. No matter how good it is, every piano has some notes that are weaker sounding than others, so always go for the best notes and avoid the clunkers.

If you have enough memory, I would suggest sampling enough notes so that no single sample is transposed up or down more than two semitones. In a perfect world with 128 MB of RAM, you could sample just about every note and have a near-perfect digital duplicate of your original piano.

If the memory in your sampler is slightly more modest, you might consider sampling fewer low notes (and at a lower sampling rate). This frees up valuable memory for notes in the mid and upper registers, which is where you will be doing most of your playing, anyway. You probably won't play so many low notes that the split points between samples become too noticeable in actual use.
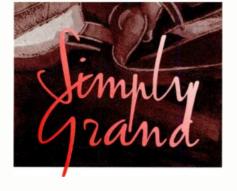
In some cases, you might find two notes right next to each other that sound great. Sample them both and keep

working your way up the keyboard. At some point, memory considerations will probably force you to choose one or the other. As I'm maxing out on memory, I usually have to sacrifice a few extra notes in the memory-hogging low range or even in the middle ranges. Alternatively, I might convert the sample rate of some low notes downward (for example, from 44.1 kHz to 32 kHz or even lower). Save the original samples to disk, though; you might eventually acquire more RAM, and you can add more of these notes as memory becomes available.

Ultimately, there is no "right" way to select which notes to sample. You might sample C2, D2, and G2, followed by B2 or even C3. The only way to determine if this is effective is—yes, you guessed it—to listen. Keep in mind that your finished piano sound will be played in a piano-like fashion, so tiny timbral shifts between samples will probably never be heard. In fact, most people would be shocked at the timbral differences between adjacent notes on some real grands.

Low notes need not be sampled at a rate of more than about 32 kHz. Indeed, except for the hardest strikes in the mid to upper registers, you rarely

FIG. 2: Placing the microphones on either side of the player position produces a realistic, stereo effect.

Simply Grand

need to use a 44.1 kHz sample rate at all. (The piano I created for Constance Demby was sampled entirely at 32 kHz and requires 12 MB of RAM.) As an experiment, try sampling a particularly bright note at 32 kHz and 44.1 kHz, and see if you can *really* hear any difference. In general, I'd rather have a longer sample at a lower rate, which is easier to loop than a shorter one that was recorded at a higher rate. The 44.1 kHz sample might have an extra 1 percent of frequency response, but only children and pets can actually hear the difference.

Always make your initial samples longer than you think you will need to allow for a good loop. I usually shoot for about eight seconds on the lowest notes, gradually shortening the samples to about three seconds for the higher octaves. That way, if you find a great loop five seconds into a low B♭, you get a nice little 3-second gift of memory that the notes arising from that B♭ don't need. However, if you only have a 5- or 6-second sample and you can't find a good loop, you might have to resample the note.

LOOP THE LOOP

Assuming you're happy with the sound of your samples up to this point, the final hurdle is looping them. I suggest that you create your finished piano by assigning individual samples to key-maps before you start looping. At this point, you should be so overjoyed with the sound that you won't mind the hideously painful headaches brought on by hours and hours of listening to buzzing loops.

If you have enough RAM, consider using fairly long samples that will rarely be held long enough to get to the looped portions. Fortunately, notes in the upper two or three octaves usually loop fairly easily because the sound evolves into something closely resembling a sine wave within a second or two. However, the lower notes are not so kind.

When a low note is struck on a piano, the vibrations of the string take a long time to settle down; they seem to evolve endlessly. Above the lowest octave or so, grand pianos have two or three strings per note. Because the strings are never perfectly in tune, there is always some harmonic motion going on.

Sometimes the best approach is to grab two or three seconds at the tail end of a sample and loop that section with an equal-power crossfade. This technique creates a subtle shifting sound that actually seems quite natural when three or four notes are held down. If your samples are six or seven seconds long in the lowest octaves, you might find a point near the end where a short, linear crossfade will work. (For more on sample looping and cross-fades, see "In the Loop" in the September 1996 **EM**.)

EASY DOES IT

Don't try to create the ultimate sampled grand piano in a day. Your ears are going to play tricks on you, and you'll end up cranky and irritable the next day when you discover you need to go back and redo a dozen samples. Spend a few hours a day, take frequent breaks, and then come back to reevaluate what you're creating. I've often been dissatisfied with a set of sounds, only to return to them the next day wondering why I was so unhappy.

I know you've heard it before, but I'll say it anyway: back up your data regularly and often (every fifteen minutes or so). Power outages tend to happen just as you're putting the finishing touches on that last G# in the highest octave.

What you should end up with is a piano that sounds like a very good recording of the piano you sampled. More importantly, you'll end up with *your* piano, the one that sounds great because *you* created it, start to finish. No commercial piano sample will ever include that as a bonus. Now take that great piano and make some great music!

FIG. 3: Try combining close miking and room miking from the audience position to capture both the impact of each note *and* the sound of the piano in an acoustic environment.

*Jim Miller is a freelance sound designer and a frequent contributor to **EM**. His samples have appeared in libraries from Sweetwater Sound, Roland, InVision, Peavey, and Kurzweil as well as in many ROM-based instruments from major manufacturers.*

Creating a Monster

A sound designer recounts life in the multimedia trenches.

By Gene Porfido

Before the multimedia craze of the 1990s, the title "sound designer" was usually associated with film, television, and theater production. Sound-effects specialists, Foley artists, and a crew of post-production professionals worked as a team to create just about every audio and ambient sound associated with your favorite movie or TV show. Imagine Arnold, pumped and primed on the *Terminator 2* set, ripping away on his Harley FatBoy, blasting the bad boys with a double-barreled shotgun that in real life probably went "phit phit phit." Sorry, Arnie, not very impressive. But by the time the gun sound has been "designed," recorded, edited, and set to film, what we see and hear in the theater is a perfect depiction of the leathered muscle man banging holes in the bad guys at a ferocious pace. *Hasta la vista,* weak little sound.
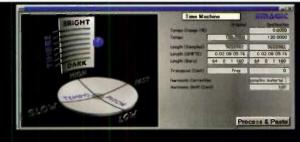
Today, sound design has spread beyond Hollywood. Technology and big business have successfully merged sound, video, music, and user interaction into the hottest trend of the decade, opening a wealth of new jobs for musicians and audio specialists. Sound design has become a crucial aspect of multimedia production as well as an extremely creative one. Sound designers set the mood and provide important audio stimulation for interactive entertainment and educational projects, much as their predecessors did for movies and television.

GETTING INTO THE GAME
Soon after multimedia drifted into my realm of experience, I desperately wanted to become a part of this awesome field. I thought the move to this megamedia would be a natural progression in my 20-year career as a guitarist, live-sound engineer, and studio engineer-producer. What could be better than continuing my career as an

With ImaginEngine's *Awesome Animated MonsterMaker,* players construct their own monsters. Shown above is Octobrat, cruising for trouble.

engineer-musician while spending hours on the Mac and getting paid for it?

By utilizing music-business connections and paying attention to the Silicon Valley grapevine, I have been fortunate enough to secure work doing music and sound design on a couple of multimedia projects. Recently, I created sound effects, atmospheres, and audio enhancements for *Awesome Animated MonsterMaker,* a game produced by ImaginEngine, a multimedia developer based in Mountain View, California. ImaginEngine, distributed by Houghton Mifflin Interactive, is known for its work in children's communications, educational products, and the popular Agent-X pen-pal network.

More than a million lines of custom-coded graphics and audio "engines" (the program/algorithm that basically *is* the game) went into *MonsterMaker.* The premise of the game is to construct crazy monsters using various characters, ingredients, and personalities culled from different rooms on a private island (owned by a mad scientist, of course). Each room or scene allows the gamer to piece together a different aspect of his or her monster; hence, each monster is unique. Because of the great variety of ingredients and the level of user interaction, gameplay is exciting and, at times, tremendously funny.

ALPHA ONE

The preliminary specifications for the *MonsterMaker* project consisted of between 300 and 400 different audio sound effects. Each section of the project was broken down on a cue list or master sound-effects list. The required sounds included various character themes, room ambiences, interstitials (music and/or effects that occur in between game actions), object- and character-movement sounds, moods, emotions, and environments. During creation of the actual CD-ROM, the cue list would be fortified with various animation files compiled in Macromedia *Director,* the industry standard for multimedia authoring that integrates audio, video, animation, and MIDI.

The first completed phase of most multimedia/software projects is called the *alpha,* which is an animated rough draft of the project. Many multimedia companies will approve a project but wait to see the alpha before they give the final go-ahead for completion of the title. For the *MonsterMaker* alpha, I needed to create about 30 sounds, just enough to provide the general idea and flavor of the final product.

Working with a storyboard (a rough visual layout of the project and its story line) and armed with a Power Mac 7100/80AV; an indestructible Mac IIci; Digidesign's Sound Tools II and *Sound*

Designer II; my trusty Casio FZ-1 sampler; Opcode's *Studio Vision Pro;* an array of electronics, outboard gear, and removable hard-disk drives; my Smith and Wesson; and a willing imagination, I began the creative design process that would ultimately end up spinning at over 3,000 RPM on some lucky kid's multimedia computer. And even though ImaginEngine is based in California, I was able to work out of my studio in Bergen County, New Jersey, communicating by phone and modem with Mark Loughridge, the project's creative director, who was at the company headquarters in Mountain View.

A lot of work for the alpha relies on imagination and insight as much as it does on computer savvy. Of course, this holds true for *all* stages of a project, but the lack of any real character development at the alpha phase makes the process both difficult and easy. It's difficult because there is really no set visual personality to score to, and it's easy because, in that respect, the project is very flexible.

The "flux" aspect of game design sets it apart from standard film/video work. A movie is usually scripted well before filming begins, so by the time the composer and effects editors get their hands on the project, the scenes are pretty much defined. However, a lot of improvisation occurs during the creation of a multimedia title. Multimedia development is extremely flexible because it's a young industry that uses constantly changing technology. There is no single way to do things, so there is plenty of room for developing any kind of breakthrough tactic that the imagination can conjure.

Believe it or not, I sent in the alpha-stage audio files on *cassette.* I could get away with the low-tech route because, at that stage in the project, sound effects only functioned to enhance the visual portions of the alpha so that the project could secure management approval and pass to the next stage of development (which is all-important if you want to keep the money flowing— and you do). In addition, the cassette format allowed the game developers to drop in the sounds easily via the Mac's standard 16-bit sound-in port.

Some developers substitute stock sounds or quick, vocalized "impressions" for alpha-phase sound effects. I prefer to deliver high quality work that lays the foundation for the final sounds.

Here is the kitchen where players choose and combine the ingredients for their monsters. Scenes such as this present the player with a lot of options, so the sound designer needs to create and layer a wide variety of sound effects.

The reason? Well, everything you create not only represents your creative abilities to the game developer but can also affect how (and if) the project proceeds. In many cases, animation work beyond the alpha stage will be governed by the sound effects because it's usually easier to animate and edit to a soundtrack. In other words, if the sound designer does a good job of visualizing game characters and environments during the alpha phase, the final art work may actually revolve around the sound files. That's good.

THE EAR OF THE BEHOLDER
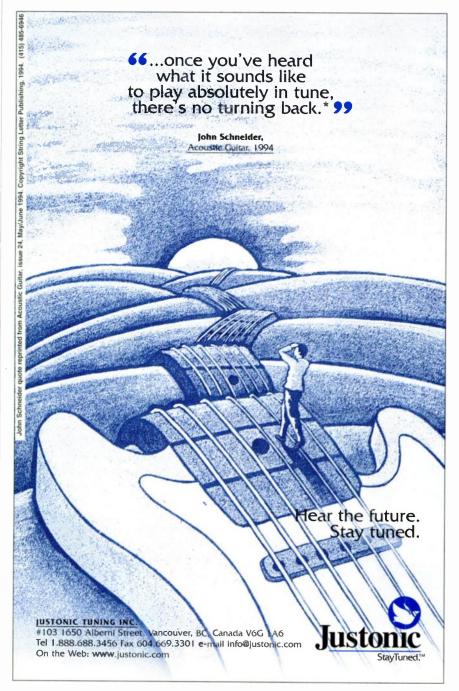
Fortunately, the alpha for *MonsterMaker* was well received, and I went ahead with the task of creating the remaining 450 or so sounds. My technique for creating the game audio entailed using two Macs. I used my Power Mac 7100 to run the *Director* animations, and the IIci, containing the Sound Tools II audio card and running *Studio Vision Pro,* was dedicated to recording sounds to hard disk. As I watched an animation file on the 7100, I first needed to figure out what the heck the animation

was supposed to sound like. Did I need to portray an emotion, a contact sound, a room ambience, or combinations of any number of effects? In addition, I wanted the soundtrack to hold up on its own so that if you closed your eyes while the game was running, you'd still have an idea of what was happening onscreen. Finally, I intended to exaggerate the sounds a bit so that they would translate well to the small multimedia speakers that are standard fare on most computer rigs.

The next process included a few different steps: deciding what the sound would actually consist of, figuring out how I was going to create the sound, and inputting the audio into my Casio FZ-1 sampler for manipulation. As with conventional songwriting, once the initial idea for a sound came forward, the rest seemed to flow and fit right into place. Then the search was on for the proper elements to create the sound. I would look for either the perfect sample from a sample-library CD or, in most cases, any available sound-making items. Believe me, your house, yard, and neighborhood are *full* of great sounds that, if manipulated properly, can create an awesome effects repertoire. Just close your eyes on a summer day with the windows open and notice all the noise that's going on—stuff you normally ignore or take for granted.

I found that I could create sounds in much the same way that professional Foley artists use props, gravel pits, and other strange objects. For example, I created spotlight on/off sounds by flicking my fingernail on drinking glasses filled with different depths and types of fluids. I mixed these samples with the sound of a flash bulb going off, and the end result was extremely authentic. For a periscope sound, I very slowly dragged a key around the threaded edge of a cheap metal flashlight. After pitching the sound up or down a bit, I had a convincing and realistic periscope. (For more on creating samples from "found sounds," see "Sample Your House and Other Fun Emax Tips" in the June 1989 **EM**.)

To use a sound from a sample CD, I usually plugged my CD player into a mixer channel (for a little EQ and other signal processing) and then into the FZ-1's ¼-inch audio inputs. For creating my own sounds, I used a Shure SM57 dynamic mic to record either to DAT or directly into the FZ-1. The FZ-1

has a great sample-edit window that lets you select waveforms by resolution; you can scale it to a single sample, and at 36,000 samples per second, that's an extremely accurate edit window. If I didn't have the FZ-1, I'd have to rely completely on numerical editing—the system used by the venerable E-mu Emax—or go the long road and transfer (digitally, if possible) the sound to my computer, edit it in a program such as Digidesign's *Sound Designer II* or Passport's *Alchemy,* and then transfer it back to a sampler so I could layer it with other samples to create the final sound. Plug-ins for *Sound Designer II,* such as *Hyperprism* from Arboretum Systems or any of the Waves products, would also let me add digital effects, compression, and/or EQ at this time. (For the full lowdown on computer-based effects processing, see "The Budget Desktop Studio" in the September 1996 issue.)

Although a single sample from a CD or other source was occasionally the *right* sound, many of the final sound effects in *MonsterMaker* are blends of two to who-knows-how-many different sounds. Creative sample layering can breathe life into sound effects and keep the audio portion of the game from becoming stale.
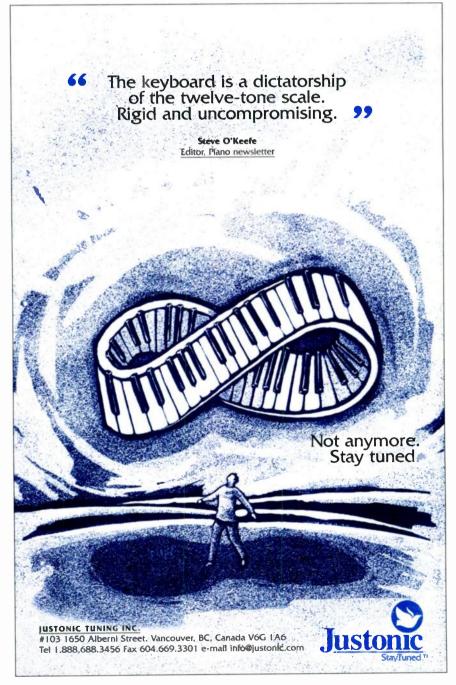
Layering chews up memory, however, and I often bumped up against the limited storage capacity of my FZ-1. For example, if I was working with an animation file of a kitchen scene, that one scene might involve 40 to 50 different actions that required specific audio effects. These effects might include a sound to call attention to a closed container (a drawer or garbage can), a container opening sound, an idle creature sound (what the character would sound like while just hangin' out), an ingredient spew sound (lots of disgusting and sloppy discharges that the kids and upper management love so much), an ingredient idle sound (while it's on the way to the boiling cauldron to form monster parts), and the sound of a container closing. Because each of these single audio effects was sometimes constructed from three to five samples layered together, available memory vanished rather quickly.

To work around the FZ-1's memory limitations, I used *Studio Vision Pro* for designing scenes of this magnitude. I'd create a new sequence for each room and program the specific sounds that occur in that room—such as a lid open-

ing—into a sub-sequence. *Studio Vision Pro*'s great editing capabilities made it a snap to do all of my sound layering, pitch adjustments, and other manipulations within the MIDI sequence. So, although an ingredient spew might consist of five sounds, I could save the layered audio effect to the FZ-1 as just one sound. Good-bye to memory-limitation frustration!

For final editing, the sounds needed to be loaded into my Mac and converted to 16-bit, stereo, 44.1 kHz *Sound Designer II* files. The project's programmers required *SDII*-format files so that they could use a custom authoring application, written expressly for multimedia standards, that would downsample a sound file to a miraculous space-saving 8-bit, 22 kHz AIFF file (Audio Interchange File Format). This killer application not only sliced a sound file to as little as one-tenth its original size, it also included a proprietary spectral enhancer that infused each AIFF file with amazing sound quality—the 22 kHz files sounded nearly as good as the original 44.1 kHz files!

The lab where the monsters emerge in their final forms.

Once each sample was opened in *Sound Designer*, I could go through each file and make final edits. Each edit was painstakingly precise, and each sound file included a very short (usually 0.05-second) fade in and a 0.5-second fade out. I used this technique to help stop pops and clicks that occur when a sample is accessed quickly by the game's playback engine. The fades are completely unnoticeable, and the smoothness of sample entrance and exit is quite an improvement over many early multimedia titles.

Finally, the files were ready to upload to the ImaginEngine team's on-line account. When I first started doing sound design for multimedia, the audio files were so large that I had to send my work to the West Coast on a 270 MB SyQuest cartridge via FedEx. Today, smaller, downsampled files make online transfers a breeze. For example, a typical upload of 50 sound effects consisted of 2 or 3 MBs of AIFF files and required a transfer time of one hour or less. I just logged on, spit out the files, logged off, and waited for that phone call or e-mail message with either congratulations or requests for slight fixes. Over the course of the six or seven months that I worked on the *Monster-Maker* project, a lot of sounds were added and a few were dropped as the project evolved and kid-testing revealed necessary changes. For the most part, I was very lucky; only about one percent of the files I sent to them needed revisions, and the development team was very pleased with my work.

LIFE'S A CARTOON

Even though sound design involves a lot of tedious and precise work, it is also a great outlet for creativity. Sound design is an important ingredient in any successful multimedia title, and the techniques used to accomplish this art are evolving every day. New technology, faster frame rates, 3D sound, and full-frequency bandwidth are just around the corner. Furthermore, tomorrow's full-screen, real-time graphics will demand more powerful audio that jumps off the screen and surrounds the user in a multimedia dream world.

With all this exciting technology, however, I still think back to all the great Warner Bros. cartoons. Although the animation is still considered by many to be unsurpassed, I'm more impressed by Carl W. Stalling's music and all the amazing voices and sound effects created by Mel Blanc and the rest of the Looney Tunes team. Without these aural treats, I know I would just stare at the screen with a lost expression on my face, wondering, "What's up, Doc?"

Gene Porfido's many projects leave him just enough time to gas the Harley and feed the pigs at his production company, Smilin' Pig Productions. When not glued to his Mac collection, he can be found by yelling "Geno!" into the local pizzeria.

Enhancing the Spectrum

Add luster to listless tracks with a bit of black-box magic.

By Brian Knave

Fifteen years ago, spectral enhancers were rare and expensive pieces of gear likely to be found only in major recording facilities and well-appointed broadcast studios. But manufacturers eventually came out with lower-cost alternatives to the original units, and today, it's not uncommon to see these specialized signal processors in home and project studios.

Unfortunately, these marvels can be very tricky to use. Applied judiciously, spectral enhancers can impart presence and sheen to individual tracks or to a complete mix. On the other hand, used indiscriminately, they can fry your tracks into crispy wisps of distortion. And the challenge is compounded by the fact that there are several different types of spectral enhancers on the market, each with different circuitry, functions, and features.

This month, we will clear up some "spectral mysteries" by revealing the ins and outs of three popular models that cost approximately $350 each: the BBE 462 Sonic Maximizer; the dbx 296 Spectral Enhancer; and the Aphex 104 Aural Exciter Type C^2 with Big Bottom. We'll discuss how these babies do what they do, what all those parameter knobs actually mean, and how enhancers are typically deployed. (For some technical background, see "From the Top: Spectral Enhancers" in the May 1993 **EM**.)
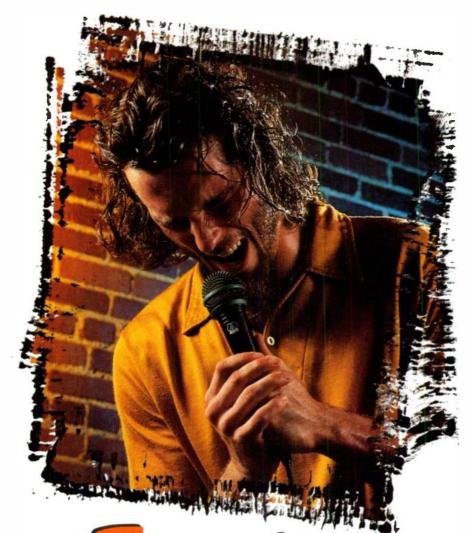
ENHANCE THIS

Spectral enhancers are commonly used at the *end* of the audio chain to process a stereo mix. This is done by patching the enhancer into the mixer's stereo bus insert or by routing the console's stereo outputs into the enhancer and the enhancer's outputs directly to the mixdown deck. But spectral enhancers are equally effective at processing individual tracks via the console channel inserts. You can also process multiple tracks—but not the entire mix—by assigning the desired tracks to a subgroup and patching the enhancer into the subgroup inserts.

In addition, you can enhance a specific effect, such as a lush hall reverb, so

By imparting a slight linear phase shift to the audio spectrum, the BBE 462 Sonic Maximizer helps extend the sense of dimensionality in a stereo mix.

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Experience the new Emotion series at your local AKG retailer and add an entirely new dimension to your performance, as Emotion delivers "The Sound That Rocks" for a lot less money than you would expect.

that it "pops" better in the mix. Simply plug the return from the effects processor into the spectral enhancer's input, and then patch the enhancer's output into the board's aux return jack (or an available channel input). If you like how enhancement clarifies the sonic details of individual instruments and effects, you may also find that it makes sense to record spectral enhancement direct to tape—especially if you own only one unit.

Recording with enhancement would allow you, for example, to process a limp keyboard pad and a muddy bass track individually, and still have the unit available to spice up the entire stereo spectrum during mixdown. A sensible way to track with enhancement is to patch the enhancer's input into the direct out of the channel you wish to process. If your console doesn't have direct outs, you can route the desired track to the processor by using an aux send. Just be sure to check all your mixer's aux send knobs to ensure that no other tracks are sent to the processor. In addition, it wouldn't hurt to solo the selected aux send and listen for any "ghost" signals hitching a ride on the sound you want to process.

It is critical that *only* the return from the enhancer—the full "wet" signal—is recorded to tape. Do not mix the dry source sound and the wet, processed signal together, because the combination may produce phase anomalies. For example, if you desire less enhancement on that limp keyboard patch, use the parameter knobs on the enhancer to decrease the effect rather than adjusting a submix balance between the keyboard and the effect return.

ALL THAT YOU CAN BBE
The BBE 462 Sonic Maximizer enhances in two steps. First, it imparts a linear phase shift across the audio spectrum. The idea, explains Paul Gagon, BBE's vice president of engineering, is "to restore harmonics to their original time arrangement.
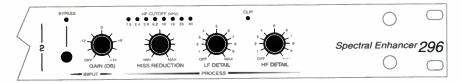
FIG. 2: Some of the high-frequency content lost to the Hiss Reduction circuit on the dbx 296 Spectral Enhancer can be restored with the HF Detail control.

"Speakers tend to slow down high harmonics," continues Gagon. "Because sound can't be sped up, the Sonic Maximizer compensates by slowing down the lower frequencies. For every cycle you go down in frequency, it progressively delays the signal by microseconds. Between 20 kHz and 20 Hz, the Sonic Maximizer imparts about two milliseconds of delay. The ear doesn't hear it as delay, though, because there's no dry signal to compare it to."

After completing the phase shift, the 462 uses a separate circuit to apply dynamic EQ to the phase-corrected signal. First, it divides the spectrum into three bands and then, while steadily analyzing the input signal, it alters the amount of boost or cut for each band accordingly. The 462 gives the user amplitude control over the low and high frequencies, but the mid frequencies are controlled internally by the unit.

The 462 is easy to use, but it has a few counterintuitive features that we'll discuss in a moment. First, let's take a look at the front panel (see **Fig. 1**). Each channel has two knobs—Lo Contour and BBE Process—and an LED display. Between the two sets of knobs is a function switch that allows you to A/B the processed and unprocessed signals.

The unit's Lo Contour knob is a low-frequency amplitude control centered at 50 Hz, with a bandwidth of approximately two and a half octaves. This control provides a 10 dB boost when turned hard right and a 10 dB cut when turned hard left. Unity-gain position (flat, with no boost or cut) is around eleven o'clock, which corresponds to the number 4 on the dial.

The Process knob adjusts the high-frequency amplitude. The filter is centered at 5 kHz and covers a bandwidth of nearly two octaves in either direction. Note, however, that the Process control doesn't cut frequencies: when turned fully counterclockwise, Process is off. Turned fully clockwise, however, Process provides 10 dB of boost. This is what I mean by counterintuitive: the Lo Contour knob is at unity-gain setting in the eleven o'clock position, while unity gain for Process is at the counterclockwise setting. Also, note that the LED displays read in opposite directions.

When using any spectral enhancer, start with flat settings. That way you can hear the process as you dial it in. With the 462, start by turning Lo Contour to 4 (the eleven o'clock position) and Process hard left. Now, when you engage the Function button, you'll hear the phase shift without the dynamic EQ. Next, check the LEDs to make sure a healthy signal is being output. According to Gagon, as long as the -20 dBu light is on, you're okay. To maximize levels, however, go for the 0 dBu setting. The Clip LED illuminates at +15 dBu to let you know that you are 3 dB below the +18 dBu clip point. But you probably won't see it light up much because the 462 has plenty of headroom.

Once the meters are showing the desired level, dial in some Process (the highs) followed by a bit of Lo Contour to fatten up the bass frequencies. Go back and forth until you reach the desired amount of equalization. For processing an entire mix, I tend to like the Lo Contour at 5 (straight up) and Process around 3 or 4. Of course, if the material is especially bright to begin with, you may not need much, if any, high boost. Or, if it's bass heavy, you can use Lo Contour to cut. When you find the levels you like, use the Function switch to A/B the processed material with the original signal.

Boosting, of course, adds level to the original signal, so keep an eye on the output levels of your board as well as
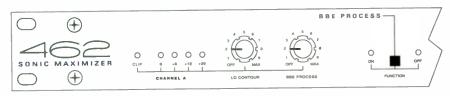
FIG. 1: When using the BBE 462 Sonic Maximizer, note that the unity-gain position for Lo Contour is around 4 on the dial whereas it is at the Off position for BBE Process.

the meters on your mixdown deck. If you know you'll be using more than a pinch of the 462's EQ on a mix, set your master fader a notch or two lower than unity to start. Also, engage Function *before* using the console EQ because the 462 will definitely alter whatever equalization you've dialed in.

DBX MARKS THE SPOT

The dbx 296 Spectral Enhancer also uses dynamic EQ to enhance signals and features independent Hiss Reduction circuitry on both channels. (BBE offers a model—the 362NR Sonic Maximizer—that also incorporates hiss reduction.) The other controls are Input Gain, Low-Frequency (LF) Detail, and High-Frequency (HF) Detail. Both channels also feature an HF Cutoff LED and Clip LED indicator. Two Bypass buttons (one for each channel) allow A/B comparisons.

The 296's Gain control (see **Fig. 2**) offers 14 dB of gain. Unity gain is in the middle (conveniently marked "0"). Hiss Reduction, which the 296 adds to the dry signal before detailing, is a single-ended noise-reduction system that

works dynamically with the HF Detail circuitry. The HF Cutoff LED display shows the bandwidth of the hiss reduction filter at every moment the process is active.

The LF Detail knob provides both boost and cut in a 2:1 inverse ratio: a maximum 12 dB of boost centered at 80 Hz and a maximum 6 dB of cut centered at 250 Hz. (The 250 Hz cut helps reduce the muddiness that can clutter midbass frequencies.) The HF Detail knob controls a dynamic filter in which amplitude, frequency, and bandwidth change constantly based on the input signal.

To set the 296, first turn down the Hiss Reduction and LF and HF Detail knobs and make sure the Bypass buttons aren't engaged. Start with the gain control at zero and increase it until the
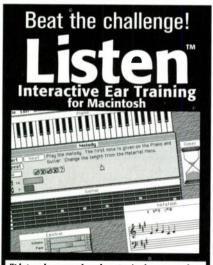
Clip LED illuminates. Then simply back off until the LED no longer lights.

Next, set Hiss Reduction where you want it. Of course, like any single-ended noise reduction system, the process chops off some high end. But the cool thing about the 296 is that the HF Detail lets you restore high-end clarity after removing the hiss. The more hiss reduction you use, the more HF Detail you need in order to compensate for the lost highs. But at the same time, the unit automatically *reduces* the amount of HF Detail available while Hiss Reduction is active. When applying it to a mix, I tend to use Hiss Reduction very sparingly, if at all. If there's no hiss or other high-frequency noise you need to diminish, don't use it.

Now gradually turn up the LF and HF Detail knobs, one at a time, while

FIG. 3: Users of the original Aphex Aural Exciter Type C² were often confused by control names such as *Girth* and *Overhang*. Fortunately, the new faceplate offers extra names that are more clear.

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The dbx 296 Spectral Enhancer uses dynamic EQ to sweeten highs and add *oomph* to bass frequencies. It features a separate hiss-reduction circuit and a high-frequency detail control that can restore some of the high-end clarity lost to hiss reduction.

listening to the results. In general, low to moderate settings give the best sound. (On a mix, I like the LF somewhere between 1 and 3 and the HF around 4, 5, or 6.) Again, if you use Hiss Reduction, try to balance the loss of highs with extra HF Detail.

The 296 also works well solely as a noise-reduction unit. Try using it during tracking, especially if the sound source is noisy. During a recent mix, I got good results using Hiss Reduction to squelch the racket from a noisy guitar amp. I then ran the HF Detail almost full blast, which gave the guitar tone lots of bite.

ALL EXCITED

The Aphex 104 Aural Exciter Type C[2] with Big Bottom is effectively two dual-channel processors in one: the Exciter processes highs and mids, and the Big Bottom covers the lows. The Exciter works by splitting an input signal, applying phase shift and synthesized harmonics to one side of it, and then summing the processed signal with the unmodified one. According to Marvin Caesar, president of Aphex, "Reproduced sound is missing some of the natural harmonic detail that's present in the original live sound. The Aural Exciter recreates and restores those lost harmonics."

The Big Bottom circuit also utilizes a split signal. Consisting of a 100 Hz low-pass filter and a dynamics processor, it basically makes a copy of incoming low-frequency information, delays and reshapes it a bit, and then adds it back to the original signal. The process is reverse amplitude dependent, meaning that more is applied as the input level drops and less as the signal gets hotter. Together, the dynamics processor

and time delay create longer-duration bass frequencies that are perceived as being louder yet don't noticeably increase peak output.

Aphex recently did a face-plate makeover for the 104 (the former unit and the new 104 are otherwise identical) and added some key descriptive words (see **Fig. 3**). This improvement, along with a revised and expanded manual,
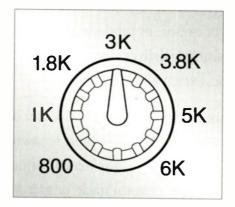
FIG. 4: The Tune control on the Aphex Aural Exciter lets you specify the center frequency of the highpass filter. The more the control is turned clockwise, the smaller the range of mid frequencies that gets enhanced.

should help demystify the names and functions of the controls.

Each Exciter channel has three controls: Tune, Mix, and Harmonics. The Tune knob sets the center frequency (from 800 Hz to 6 kHz) for the range of frequencies being enhanced by the Exciter (see **Fig. 4**), and Mix determines how much of the enhanced signal is mixed with the unmodified signal. The Harmonics button is labeled Normal and High. The word *high* refers to the level, not the frequency, of the harmonics in the overall Exciter process-

ing. Pushing the button to the High setting provides an additional 9 dB of harmonic content.

Big Bottom has two controls per channel: Overhang/Drive and Girth/Mix. Overhang sets the input level and Girth determines how much enhanced signal is mixed with the unmodified signal. Note that the Harmonics switch affects the Exciter circuit only, not Big Bottom. The Process button engages the unit when depressed and works as a bypass in the out position.

Because the Big Bottom effect is more noticeable than the Exciter, it's best to start with the Exciter when setting the 104. First, set the Exciter's Mix control to maximum (fully clockwise). Then vary the Tune control and listen for the frequencies being enhanced. For general enhancement, set it in the middle; for more air, try two or three o'clock. After locating the desired frequency setting, dial the Mix control back to zero, and then start slowly adding it again. When you can clearly hear the enhancement, turn it back down a bit.

The key to successful use of the 104 is moderation. "A little bit goes a long way," explains Caesar. "If the mix sounds 'effected,' you've used too much. The sound should be noticed more in its absence than its presence. When you switch Process off, everything should take a slight step back into the speakers. There shouldn't be a gross spectral change."

Of course, that's for a complete mix. On an individual track, you may want to hear a more enhanced signal. The Aural Exciter is particularly good at highlighting the sound of percussion and sound effects without making them louder in the mix. For more enhancement—say, to give a snare drum extra bite—push in the Harmonics button to the High setting and readjust the Mix control accordingly.

To set Big Bottom, first turn the Girth control all the way down. Next, add input with the Overhang knob until the LED flashes steadily on the peak bass notes. That's the correct input level.

The Aphex 104 Aural Exciter Type C[2] with Big Bottom analyzes the audio spectrum and generates missing harmonics that help "excite" the sound. The Big Bottom circuit uses delay to subtly fatten bass sounds without adding appreciable gain.

Now slowly turn up Girth (Mix) until it feels as if the bass player took a step forward. Be careful not to mix in too much, though, or the sound will get muddy. All you want is a subtle increase in bass presence and detail. A Girth setting of eleven or twelve o'clock is usually the most you'll need. The bass notes should sound fatter but not louder.

Note that the 104 has operating-level switches (one for each channel) on the back panel. Naturally, these should be set to match your console's operating level. The problem is, some mixer manuals don't divulge the operating level of

the board's channel inserts. Therefore, use common sense. If the Overhang LED isn't flashing by the time you reach two o'clock on the dial, you're probably on the wrong setting, so switch the levels to -10 dBV. But if it's flashing madly while the control is barely on, switch the levels to the +4 dBu setting.

FINISHING TOUCH

Just as a car detailer can transform a dirty, neglected vehicle into an eye-catching prize, spectral enhancers can add the finishing touch that makes a track sparkle and shine. But proceed with caution: the immediate hit from an enhancer can be seductive and has lured many an engineer into enhancement overdose. The trick is restraint. Find the settings you like, then reduce them a tad. Remember that the point is to *enhance* the signal, not to fry it.

Assistant Editor **Brian Knave** *wishes for a box that would enhance the level of political discourse in the U.S. (Special thanks to Mike Charles, Marvin Caesar, Denny McLane, Paul Gagon, and Rob Rizzuto.)*

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❏ For impedance matching, recording direct, running an instrument direct to a mixing board, or for warming up a digital mix, we recommend the ADB3 Stereo Active Direct Box. It has two channels, each with a ground lift, +/-20 dB of gain, a 20 dB pad, and transformer coupled XLR outputs. Only $200 retail.

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Hybrid Synth Power-Up

Hot troubleshooting tips for keeping the juice flowing.

By Alan Gary Campbell

When you throw the switch on a hybrid synth, a lot of things happen in a matter of seconds, but they have to happen in the correct mode and sequence or the instrument will "wake up" very confused. When the power comes on, the raw DC power sections of the various analog and digital power supplies start to charge up. The supply capacitors draw a lot of current for a moment, and then the DC supply sections stabilize but not all at the same rate. Some supply sections are designed to provide less current than others and have smaller main filter caps that charge more quickly.

As soon as the minimum acceptable raw-DC voltage is obtained, the power supply regulators kick in and attempt to bring the various supply lines to spec. You'll often get glitches and noise on the lines as the raw-DC inputs settle.

When the regulators kick in, digital and analog circuits, left alone, try to start simultaneously and randomly. Typically, the digital clock starts to run; the microprocessor and other "intelligent" ICs wake up and try to process data; the analog oscillators, filters, and envelope generators start; and so on.

This would be cacophonous chaos if not for the reset circuit, which establishes an orderly power-up sequence. A common reset-circuit design uses inverters and passive components to form a Schmitt Trigger, another common design uses a 555 Timer in one-shot mode, and some designs are entirely passive. In each case, the reset circuit is configured so that its trigger cycle begins as soon as the +5-volt digital supply starts to come up. The reset circuit pulls the reset line of the microprocessor down briefly and then releases the line. In some designs, the reset circuit also controls an output muting circuit (remember those analog oscillators trying to power up without proper control?), which may drive an output relay, a group of output-muting transistors, or a subcircuit that closes the voice-output VCAs or a master VCA.

When the reset line is released, the microprocessor wakes up the rest of the circuitry in order. The microprocessor, of course, has to be provided with a power-up instruction set, which is generally stored in the base area of

the program ROM (or ROMs). The power-up instructions tell the microprocessor how and in what sequence to set the initial parameters for the D/A and A/D converters and what data to fetch to set the panel displays and controls and establish the power-up *patch* or program. Typically, a synth powers up with program 1, program 0, or the program or edit that was active when the unit was last powered down.

In theory, this all seems straightforward enough. But in practice, hundreds of instructions are required just to get a hybrid synth to power up in a predictable, useful state. Consider the latches that control the panel switches and the sample-holds that grab the multiplexed control-voltage output. In the Memorymoog, there are 92 such basic panel and interface functions, each requiring several instructions to fetch the relevant data and set up the ancillary chips that the microprocessor controls. That's a lot of latches and sample-holds!

POWER-UP DEFECTS
Problems involving the power supplies, clock, reset circuit, or memory can prevent an instrument from powering up properly. With a dead or "scrambled" unit, *these circuits should be checked first.*

Possible power-supply problems are especially critical because a malfunctioning supply can cause significant damage to an instrument. For example, if the positive bipolar supply is dead but the negative bipolar supply is still active, in a very short time the majority of CMOS switches and DMUX chips on the supply will be history.

If a power-supply defect is suspected, it is vitally important, during the initial troubleshooting, to run the unit only long enough to verify the circuit parameters. If the supply checks out, general troubleshooting can proceed. Otherwise, the supply should be disconnected from the rest of the circuitry and tested separately until the problem is corrected.

By far the most common failure mode for a standard IC-regulator-type power supply is a damaged or open rectifier diode in the raw-DC input section. Usually, a line transient will exceed the surge current or Peak Inverse Voltage (PIV) capacity of the diode and destroy it, sometimes with visible damage. Occasionally, the body of the diode is utterly vaporized. Such transients are usually too fast to blow a fuse or trip a breaker.

Quite often, it is sufficient to simply replace the damaged diode, but nearby components should be checked carefully for indications of collateral damage. Electrolytic caps are vulnerable; check for signs of arcing or discolored PC pads and solder joints, and check radial caps for distention of the vinyl top cover, a sign that the cap is damaged.

The 3-terminal IC regulators (such as the 7805 and its larger cousins) that are typically used in hybrid power supplies rarely fail. These devices have built-in thermal- and current-overload protection. More commonly, current-amplifier transistors used in conjunction with the regulators fail.

Much more common than a power-supply failure is a simple, usually benign, memory glitch. Although the power-up instruction set is contained in ROM, all parameters that change or are programmable are stored in battery-backed, static RAM. A line transient too small to cause any power-supply damage can still scramble the RAM contents.

If the data is scrambled, the unit can easily be "locked up" or even appear dead at the end of the power-up routine. Sometimes, the memory is so confused that reloading the data from cassette or other storage device won't work. In this case, it is necessary to manually reset the RAM by shorting the battery-backup voltage to ground for a moment. A ground-shunt lead is used for this purpose. A ground shunt can be constructed from an old test probe, a resistor, and an alligator clip (see **Fig. 1**). The probe lead is cut at a convenient point, and the 1 kΩ resistor is wired in series in-line and covered with heatshrink tubing. The connector on the other end of the lead is cut off, and the alligator clip is soldered on. To reset the RAM, power down the unit, connect the alligator clip to digital ground at an accessible point, and then briefly touch the probe to the battery's positive terminal.

A weak backup battery can also scramble the memory. A measured terminal voltage more than 10 percent to 15 percent below the rating is suspect. The terminal voltage should be measured only with a high-impedance, digital voltmeter or digital multimeter, never with a standard, analog VOM. Older analog meters have relatively low input impedances and can deplete the battery during measurement.
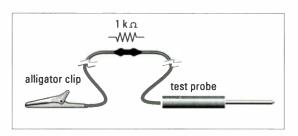
Even if the backup battery fails, the unit is still functional once it is powered up and reprogrammed because the battery backup circuit "hands off" the RAM supply line to the +5-volt digital supply during the power-up routine. Of course, if the battery is depleted, the memory will again be scrambled when the unit is powered down, but as long as the unit stays on and there are no significant line transients, it will work. And that will be enough to get you through a gig, in an emergency—provided you remembered to back up your data!

The clock circuits in hybrid synths rarely fail. These generally operate at under 20 MHz and are easily checked for signal quality and approximate frequency with a service-grade oscilloscope. It's not uncommon to see quite a bit of "ringing" on the waveform, an artifact caused by the 'scope probe. This is usually not an indication of component failure except in severe cases.

Reset circuits are problematic to troubleshoot with a standard 'scope because they "go low" quickly but briefly after power-up. A multiple-trace memory 'scope is handy to compare the reset-circuit output with the digital supply and with any secondary reset lines. Failing that, you can attempt a manual reset by using your ground shunt to pull down the microprocessor's reset line briefly. If that resets the unit, replace the active components of the reset circuit or rebuild the circuit.

EM *Contributing Editor* **Alan Gary Campbell** *is editor and publisher of the* New Music Journal *and owner of Musitech, a firm specializing in electronic musical-instrument design, modification, and service.*

FIG. 1: Stubborn memory glitches or defective reset circuits can be cleared manually with this easily wired ground shunt.

Executive Ears

Music-industry execs discuss what they listen for in your demos.

By Michael A. Aczon

In recording studios around the world, artists slave over take after take, rewriting, performing, editing, and meticulously crafting their work into demos. Artists trade these precious jewels for the rewards that come with music-industry deals. But before these demos can be exchanged for fame, fortune, and aggressive distribution and promotion, they must get past the gatekeepers to the Promised Land, those recording-industry executives who listen to music, evaluate talent, and make the deals.

In an effort to demystify the demo-shopping ordeal, I interviewed a number of industry executives to get their candid thoughts on their listening and selection processes. I have compiled some of their insights into a "round table" discussion.

The executives I spoke with work in a variety of capacities at companies throughout the music industry, from music publishing firms to record labels to performing-rights societies. They are David Hirshland, vice president of Business and Legal Affairs for Bug Music publishing in Hollywood; Ron Sobel, ASCAP's vice president in Los Angeles; Ian Steaman, A&R director for New York City–based Tommy Boy Records; Janet Parr, creative director for the Nashville-based music-publishing house Milsap/Galbraith Music; Shari Saba, senior director of Creative Services for Warner/Chappell Music publishing in Los Angeles; James Leach, a Los Angeles–based independent publisher and A&R consultant; and Kevin Walden, general manager of Perfection Light Productions, the Marin County, California, production company of multi-platinum producer Narada Michael Walden.

How do artist packages get to you? What does a "solicited submission" mean?

DAVID HIRSHLAND: The Bug version of a solicited submission is that it comes to us through channels we know, usually a manager, attorney, or writer we

work with. Occasionally we will meet a new writer face to face. An unsolicited submission, on the other hand, means that there was no previous contact with us. However, we do answer the phone daily and try hard to get back to everyone who calls here, so a person could make contact with us that way.

Keep in mind that Bug does publishing administration, so if a writer calls us who doesn't either have a record out or have a song involved with television or film that needs administration of the copyright, there's really no reason to talk to us.

RON SOBEL: ASCAP has totally open access. The songwriter or a writer/publisher rep contacts me by telephone, by mail, or by sending a tape. There may be a delay in response time because of the volume of submissions and mail and phone inquiries as well as personal meetings, but certainly we are open. Mailing a tape is probably the best way to get a demo to me, but it usually takes me a while to get to it.
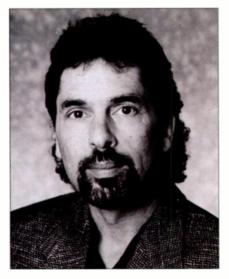
IAN STEAMAN: Well, for starters, I don't want to give you the impression that we have a company line. Everyone here, as at most labels, has his or her own style of doing things. At this moment, we're not taking unsolicited material, meaning material that's not from a source we're familiar with. If Joe Blow from Peoria with no connection to us were to just send a tape, he has to realize that I have a zillion tapes here, and I can't get to them all because there's simply too much going on. That's not to say it isn't good, I just need to get it from a source. A good way to get a demo to me is through an attorney, a manager, a production company, or an artist that I have a relationship with. I know I will get a certain level of quality, both artistic- and production-wise, from a known camp.

JANET PARR: Historically, Nashville has been a much more open market for unsigned talent. However, the recent success of crossover songs and the popularity of Nashville-based music have resulted in a lot of people coming here and looking for a break. It's so busy now, I have to get a referral before I'll listen to a tape. There's no way I can spend the time listening to every tape sent to me. I'm busy keeping up with recent releases, reviewing albums

for pitching purposes, and meeting with writers I'm already working with. This is a small town, however, with a strong sense of community, so writers I'm already familiar with turn me on to new writers; other publishing companies will turn me on to stuff; BMI, ASCAP, attorneys, and managers all refer writers to me.

SHARI SABA: I approach the evaluation of a writer or artist from an artistic and a business perspective. I was a singer-songwriter signed to Criterion a number of years ago, before I started working for Warner/Chappell, where I have been for twelve years. Everyone here has his or her own version of what solicited material is, but "solicited" in my book is a submission that has a referral attached to it, preferably from a manager or a lawyer who has a good reputation and ideally from someone who knows me. A lot of times, the referral knows your name but not much else. A good referral knows your taste, knows what you like. There are legal reasons, too, for only taking tapes from people I know. If you can't track where a tape came from, it could result in some problems down the road.

I will take tapes depending on the initial situation or conversation with an artist, writer, or their representative because you never know what you're going to hear. Even if I were to meet someone in a club or at a seminar, I prefer to have them send it on a follow-up. Again, for possible deal and legal purposes, it is much better if I can track where the tape came from.

Ron Sobel, vice president of ASCAP.

How do you listen? What are you looking for? Who evaluates the tapes?

SOBEL: For a song demo, I prefer a voice with a single acoustic guitar or a voice with a piano. I'm primarily looking for lyrics plus melody, and the lyrics have to have a point of view that is unique, meaningful, and strong. The melody also has to be unique, inventive, and challenging. Anything that is either lyrically or melodically "safe" simply wouldn't qualify, so there would be no reason to reward it or pursue it. Production is not important to me when it comes to looking for songwriters. As an artist moves up the food chain and is looking for a recording contract, all of the elements of production and instrumentation come into play, but that's certainly not what I am looking for in songwriters.

PARR: The music has to speak for itself. Having someone with me helps me get through a tape sometimes; we'll end the day by listening to tapes together at the office and batting comments around. I get my best listening in while I'm multitasking. When I let my subconscious take over, the best stuff presents itself. I guess it's because when I was little, I used to run out, buy an album, and immediately go home and listen to it while doing a jigsaw puzzle. I would invariably be able to pick all of the eventual singles because they would make me stop working on the puzzle and listen.

JAMES LEACH: I like to listen while riding in my car. Sometimes I let people who have nothing to do with the music business hear it and see how they react. Eventually, these types of listeners will be the song's ultimate test. When I go to some industry events, the artist-writer is performing songs, and that is a good way to hear the music before actually following up with the person doing the submission.

SABA: My listening habits for artist demos are different than for songwriting demos. If I'm evaluating a band, I prefer to be in my car. That way, I can listen to the act's presentation as an entire body of work rather than breaking up the listening process by taking phone calls and going to meetings. When evaluating a songwriter, I can usually do it in the office because I can

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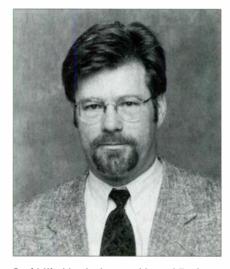
HIRSHLAND: I try to listen to everything. The first time through, I'll usually listen in my office. Sometimes I will listen in my car. We have creative departments in Los Angeles, Nashville, and London, and I try to get a second opinion from those people because they ultimately have to work the songs.

STEAMAN: If I think a project is something Tommy Boy should get behind, we do have a committee, but it's not made up just of A&R people. Every other week, the sales, video, business affairs, and promotion departments all have a say in the groups we're going to sign. I guess you can say it's by committee from the standpoint that we want to evaluate it from all sides. For example, a group might have a great sound, but it doesn't translate to video, so we all get together to figure out whether it's something that will work for all of us. We want the entire label to get behind it. A salesperson here has to be with the artist from the beginning, and having sales involved from the signing process is, in the long run, much better for the artist. Everyone has to understand the vision from the start in order for the project to work.
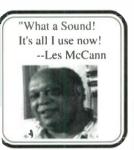
How about follow-up? Do you do it, or does the artist or the artist's rep? When are they stepping over the line and being too aggressive?

SABA: They are stepping over the line when they call me every couple of days to see whether I got to the tape yet. A lot of people don't realize that in order for us to really stay on top of

David Hirshland, vice president of Business and Legal Affairs for Bug Music.

the music and the artists, a lot more goes on than just listening and evaluating. We have to deal with paperwork and troubleshooting. I know how hard it is to be on the other side, waiting to see whether someone has gotten to your music yet. If you haven't heard anything in three to four weeks, go

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● WORKING MUSICIAN

ahead and follow up, but don't call every couple of days.

PARR: If I like it, I'll call you. If I take a meeting with you, I'll call or you can follow up. If I have specific feedback, it usually won't happen for about two weeks. If I can't help the artist at all, I try to close things at the meeting or shortly after. Treat a meeting or submission like a job interview; send a note within a week after they've listened. If

▼

Most execs won't accept material from an unknown source.

they tell you that they will be making some sort of a decision, follow up after that. I consider it a bad sign if a person calls two to three times a day. If someone acts desperate, it's a sign that they're not confident—and confidence counts as much as talent in this business. Calling someone over and over again does not make them like your songs any more than they already do.

KEVIN WALDEN: If someone takes the care to prepare a well-produced and well-packaged tape and takes the steps to get it directly to one of us, it deserves the time of one of our staff to at least listen to it. Somebody who is diligent enough to prepare a tape and get it to us usually is pretty good at following up without being a pest about it. Be a salesperson, but use tact. Take a tip from the guy in the movie Wall Street: befriend the person at the front desk and you'll eventually get the answer you need, rather than leaving a thousand messages about how mad you are because no one will call you back.

STEAMAN: Again, there's so much going on, and there are so many tapes, I don't call people back personally with rejections. If they need to hear something, they usually find a way to get back to me. If we have the relationship that I told you about earlier, I can let them know why we passed, and it's usually cool. It doesn't stop them from getting other stuff to me.

And finally, what do you do if you like what you hear?

SOBEL: For bands that I've liked, I've not only spent an immense amount of time working with the writers themselves, but I've worked within the industry, presenting tapes to publishers, A&R people, and attorneys. I am very active and aggressive in presenting a project that I think is strong.

SABA: I signed 16 Horsepower on A&M and Imperial Drag on Work/Sony, and I'm proud of those. Even though some of the projects I love are left of center, I'm really lucky, because if I love the music, I get to go with it.

LEACH: If I'm looking at the person as a songwriter-producer, I try to hook them up with a publishing deal or with acts looking for tunes. If I can't do something for them, I might refer the project to another publisher who has similar taste. On the artist side, I will shop the project around for an artist situation if other factors, like their management setup and having their heads on straight, are good.

PARR: I just signed a writer named Randi Michaels, and I'm proud of that signing. When I found her, I knew I had to sign her. I lobbied to my boss to spend money on this writer. I spend a lot of time introducing writers to collaboration situations. I try to get them in on an artist's career early because if they are in on early hits, their entire catalog becomes valuable down the road. Sometimes I'll even pitch songs before a deal is signed because I believe in good faith in the music industry. I'm a publisher, not just a song plugger or cheerleader, and I'd like to publish Randi's career, not just her songs.

LISTEN UP

These industry insiders have distilled their years of experience into valuable insights. And although these pros have different roles within the industry, they all echoed the same themes when explaining how to approach the process of shopping your demo. If you take anything away from this discussion, take these four points. First, know what kind of presentation you are trying to make. Are you an artist? A songwriter? A producer? Prepare your presentation accordingly. Second, network, network,

network your way to someone who knows the inside player you are trying to get to. Attorneys, managers, and people who are working within the organization you are interested in seem to be the best avenues. Third, have respect for the time constraints your listener is under. And finally, do not lose faith for a minute or stop working at making your talents shine.

I would like to take this opportunity to thank all of the executives who so graciously shared their thoughts with me. Bear in mind that their participation and candor in this discussion are not an open door for submissions (e.g., "I read what you said in the *Electronic Musician* article, does that mean my tape is considered solicited?"), but I'm sure that if you stay on your path and plan for success, all roads lead to these and other executives who will make your artistic dream happen.

When not practicing law and teaching music-business courses in the San Francisco Bay Area, **Michael A. Aczon** *listens to demo tapes with his wife and two kids.*

Caveat Connector

Let the user beware when sending digital audio from here to there.

By Scott Wilkinson

As you probably know by now, digital audio offers several distinct advantages over the analog variety. For example, there is no generational loss when making copies, and it's relatively easy to edit digital audio with a computer. (See "Square One: Digging into Digital Audio" in the February 1996 **EM** for more on the basic concepts of digital audio.)

As a result, many engineers want to keep audio signals in the digital domain as they pass from one device to another. However, connecting a digital audio output to a corresponding input is not always as straightforward as performing the same task with analog signals (see "From The Top: Basic Audio Connections" in the October and November 1993 **EM**).

Digital audio signals can be passed from one device to another in any of several formats. These formats define variables such as electrical characteristics, physical connectors, how the digital audio is encoded, the rate at which data is transferred, the bit resolution and sample rate, and extra housekeeping and error-detection information. Unfortunately, different formats are generally incompatible, although it is often possible to convert from one to another.
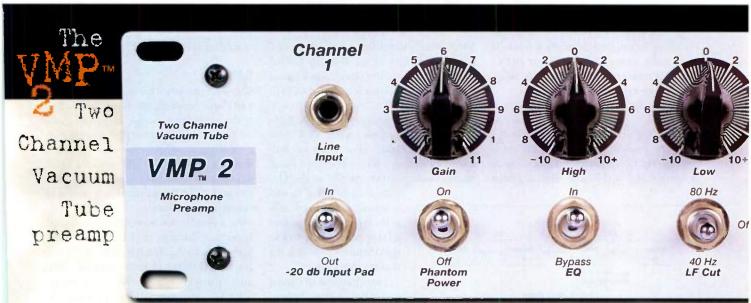
With electrically conductive cables, digital audio signals are transmitted as a series of quickly alternating voltages, which represent the 0s and 1s of the signal. With fiber-optic cables, the bits are represented by pulses of light from a laser or photo-diode. In addition, two or more separate channels of audio information are often sent along the same cable; this is called a *serial* interface. In some cases, each channel of audio is sent along its own wire; this is known as a *parallel* interface.

TIMING IS EVERYTHING

Digital audio signals are extremely sensitive to timing. Each word of data must arrive at its destination at precisely the correct moment. (A word of data includes sixteen or more bits and represents one measurement of the instantaneous amplitude of the audio signal at a particular moment in time. The entire signal is represented by many data words, typically 44,100 or 48,000 per second.) If the timing is off,

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The VMP™ 2 Two Channel Vacuum Tube preamp

Two Channel Vacuum Tube

VMP™ 2

Microphone Preamp

Channel 1

Line Input

In
Out
-20 db Input Pad

Gain

On
Off
Phantom Power

High

In
Bypass
EQ

Low

80 Hz

40 Hz
LF Cut

The VMP™ 2 gives you three tubes for every one found in our competitors' preamps. While others create the illusion of being a tube preamp, in reality most only provide a single tube in the front end, leaving a cold transistor back end to do all the work. The VMP-2's all-tube preamp gives you true tube texture because we use tubes EXCLUSIVELY throughout the signal path. And high-voltage power rails utilize 100% of each tube's energy for extended warmth—something you can't experience with a preamp powered by a wall wart. Many consider the VMP-2 a two-for-one product because of the built-in all-tube EQ, which allows the artist to deliver crisp highs or robust lows to any track or mix. Odds are, you'll love the VMP-2: the vacuum-tube power tool for the professional and the professional-at-heart.

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even by a microsecond, the final sound will be distorted when the words are converted into an analog audio signal.

The timing of each word is determined by a *clock signal*. The most common type of clock signal is called a *word clock*. This word-clock signal jumps between two values at a frequency equal to the sample rate of the digital audio data. For example, if the sample rate is 44.1 kHz, the word clock oscillates at 44.1 kHz. Each cycle of the word-clock signal corresponds to one word of digital audio data, which is why it's called a word clock.

The consistency of the word-clock frequency is critical for high-quality digital audio transfers. If each clock cycle does not occur precisely when it's supposed to, the result is called *jitter*. In this situation, the exact duration of each clock cycle is unpredictable to a certain degree, which adversely affects the quality of the signal.

All digital audio devices have their own internal clocks that regulate the timing of their own signals. If you want to transfer digital audio from one device to another, their clocks must be synchronized. This process is similar in principle to synchronizing performers using a metronome or click track. It is not the same as SMPTE or other forms of time code, which are not nearly accurate enough for this task.

There are two ways to synchronize the clocks in several devices. If you are sending digital audio from a single device, the easiest approach is *self clocking* (see **Fig. 1**). The sender's clock signal is embedded within the digital audio data itself, and the receiver syncs its clock to this signal.

On the other hand, if you want to send data from several sources simultaneously, perhaps through a digital mixer, you must use a technique called *master clocking* (see **Fig. 1**). In this process, one device's clock signal is sent from its dedicated word-clock output to a dedicated word-clock input on all other devices over a separate cable, which typically terminates with a BNC connector (see **Fig. 2**). This is similar to the *house sync* found in most video-production facilities, in which a dummy video signal is generated by a master source and distributed throughout the facility to synchronize the video equipment.

AES/EBU

In 1981, the Audio Engineering Society (AES) and the European Broadcasting Union (EBU) began working on a standard format for stereo digital audio signals. These discussions resulted in a format commonly called AES/EBU. (Its official name is AES3-1985, which identifies when it was finished.) This format was also sanctioned by the Electronic Industries Association of Japan (EIAJ), which called it CP-340 Type I.

According to the standard, AES/EBU uses balanced cables with XLR connectors (see **Fig. 3**). (For more on balanced cables, see "From the Top: Basic Audio Connections, Part 1" in the October 1993 **EM**.) It was originally designed as a self-clocking system, but a later addendum to the specification provides for master clocking.

Two channels of digital audio data are *multiplexed* on a single conductor within the cable. This means that one word of data from channel 1 is followed by one word of data from channel 2, which is followed by one word from channel 1, and so on. The resolution of the audio data words can be up to 24 bits, but most digital audio currently uses only 16 bits, which

FIG. 2: Most master-clock cables use BNC connectors. (Courtesy Neutrik)

provides for future enhancements in digital audio resolution.

The data is transmitted at a bit rate that is 64 times the digital audio's sample rate. Thus, if the sample rate is 48 kHz, the AES/EBU data transmission rate is over three million bits (megabits) per second (3 Mbps). This makes it possible to send two channels of 24-bit audio plus the extra housekeeping and error-detection bits and play the resulting stereo signal in real time.

S/PDIF

Many engineers who worked on AES/EBU also worked for Sony or Philips at the same time. Both companies wanted to develop a standard for transmitting digital audio between consumer devices, such as CD players and DAT machines. (At that time, DAT was thought to be the next major consumer recording medium.)

As a result, the Sony/Philips Digital Interface Format (S/PDIF) is almost identical to AES/EBU. The primary physical difference is the type of cable and connector used for this format. S/PDIF uses unbalanced, coaxial (or "coax") cable and RCA connectors (see **Fig. 4**), which are much less expensive than balanced cables and XLR connectors. Other differences include the way digital audio data and the auxiliary information is encoded. This format was also sanctioned by the EIAJ and called CP-340 Type II.
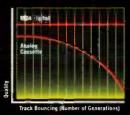
In addition to its electrical version, S/PDIF can also be implemented with fiber-optic cable made of plastic or glass and a connector called *Toslink*. This interface can be found on several

FIG. 1: In a self-clocked system (top), the word-clock signal from the sender is embedded in the digital audio datastream and used by the receiver(s) to sync their clocks. In a master-clocked system (bottom), the word-clock signal from the master device is sent to all other devices in the system on a separate cable.

FIG. 3: AES/EBU digital audio uses XLR connectors with balanced cables. (Courtesy Neutrik)

devices, including many consumer audio products and the Kurzweil K2000 and K2500 sampling inputs. Optical interfaces avoid all problems associated with electrical connections, such as cable capacitance and grounding issues. The Fostex COP-1 converts between optical and coax S/PDIF.

It is possible to convert AES/EBU into S/PDIF and vice versa. This normally requires a *format converter,* such as the Digital Domain FCN-1. Some digital audio devices, such as the Lexicon Model 300 digital reverb, include an internal format converter, which lets them deal with digital audio in either format.

Some manufacturers have implemented software-switchable AES/EBU and S/PDIF interfacing on a single connector, which can be XLR, RCA, or even a nonstandard connector such as ¼-inch. In such cases, you might need to use an adapter.

Some manufacturers have implemented AES/EBU or S/PDIF only partially or even incorrectly. As a result, some gear can't deal with a converted format or even the same format from another device. If one device implements a subset of the spec and another device implements a different subset, they might not work well together. This is one of the primary pitfalls of digital audio interfacing.

MULTICHANNEL FORMATS

When Alesis introduced the ADAT modular digital multitrack (MDM) tape recorder in 1992, they had a huge hit on their hands (and they threw a great party at the NAMM show that year!). As you probably know, the ADAT and the new ADAT XT record eight tracks of digital audio on S-VHS videocassettes. Among its features is an optical,

self-clocking digital audio interface called the Lightpipe (see **Fig. 5**).

The Lightpipe interface is serial, multiplexing eight channels of digital audio on a single fiber-optic cable with a proprietary connector. It can accommodate a resolution of 24 bits, even though the ADAT currently records 16-bit audio. Thanks to forward-thinking designers, this interface will avoid obsolescence and remain useful for future products with higher resolution. The data-transmission rate is 256 times the sample rate, which is four times the data rate of AES/EBU and S/PDIF. This relationship between data rates makes sense, because the Lightpipe

FIG. 4: S/PDIF signals use RCA connectors with unbalanced cables. (Courtesy Neutrik)

● **SQUARE ONE**

carries four times as much data as AES/EBU or S/PDIF (eight channels instead of two).

Among the growing number of products that include a Lightpipe interface is the Alesis QuadraSynth line, which sends audio from the synthesizer directly to an ADAT without leaving the digital domain. Other products with a Lightpipe interface include the ADAT clones from Fostex and Panasonic, and it is an option for the Yamaha 02R digital mixer and for dedicated hard-disk recorders such as E-mu's Darwin and Roland's DM-800. The Alesis AI-1 converts any two consecutive channels of data between the Lightpipe and AES/EBU or S/PDIF.

Shortly after the ADAT was introduced, TASCAM answered the 8-track MDM challenge with the DA-88. The multichannel digital audio interface used on the DA-88 (and its newer sibling, the DA-38) is called TEAC Digital Interface Format (TDIF). This master-clocked format carries eight channels of digital audio on an electrical cable with 25-pin D-sub connectors, which are used on many computers. Each wire in the cable carries two multiplexed channels, which closely resembles AES/EBU. The entire cable can handle eight channels to and from any compatible device; in other words, TDIF is *bidirectional*. The maximum resolution is 24 bits, although the DA-88/38 records with 16-bit resolution. In addition, the data rate is the same as AES/EBU.

Like the Lightpipe interface, TDIF is a standard or optional feature on a variety of products, including the Yamaha 02R digital mixer. The TASCAM IF-88AE converts all eight DA-88/38 channels into AES/EBU on XLR connectors and any two consecutive channels to S/PDIF on RCA connectors.

There are several other multichannel digital audio interfaces that are used mainly on high-end professional equipment. The second-generation of the Sony Digital Interface Format (SDIF2) was the first multichannel format, developed in 1981. This master-clocked, parallel format can accommodate up to 24 channels of 20-bit digital audio.

One year later, Mitsubishi introduced its Melco (Mitsubishi Electric Company) multichannel interface, which is also known as ProDigi after the recording format used by Mit-subishi and Otari digital audio tape recorders. This parallel, master-clocked system carries up to 32 channels of 16-bit digital audio. There are three versions of Melco/ProDigi, called Dub-A, Dub-B, and Dub-C.

Yamaha introduced its Y2 proprietary digital interface in 1988 as a method of cascading (chaining) two or more DMP7 digital mixers. Later, the company expanded Y2's applications to include several other Yamaha digital

FIG. 5: The Alesis ADAT uses a proprietary optical interface called the Lightpipe.

audio products. Y2 is a parallel, master-clocked format that carries one channel of 24-bit digital audio per line. Its flexible architecture can be configured to carry almost any number of channels; a cable with 25-pin D-sub connectors can carry eight channels of digital audio bidirectionally.

The Multichannel Audio Digital Interface (MADI) is essentially a multichannel version of AES/EBU developed in 1991 by Sony, SSL, Neve, and Mitsubishi for high-end digital mixers and multitrack recorders. MADI multiplexes up to 56 channels of AES/EBU digital audio onto a single coax cable with BNC connectors. This master-clocked system operates at a fixed data rate of 100 Mbps.

Clearly, sending digital audio signals from one device to another is not trivial. However, a little knowledge goes a long way toward making a successful connection. Of course, experience is the best teacher, so try making a few digital audio connections of your own. You'll probably find that the benefits of staying in the digital domain far outweigh any difficulties.

EM *Technical Editor* **Scott Wilkinson** *enjoys visiting the digital domain.*

Reviews

TASCAM DA-38

By Brian Knave

A DA-series MDM priced for the home recordist.

One thing you learn fast in this business is that product numbers often have no truck with logic. It would be logical to assume, for example, that TASCAM's new DA-38 modular digital multitrack recorder is somehow lower in stature than its predecessor, the DA-88. But in this case, the numbers are pretty much meaningless. With its improved converters and surprising array of features (several of which don't appear on the DA-88), the DA-38 is less the DA-88's little sibling than its fresh-faced first cousin. To ensure that the two machines don't get caught up in something like sibling rivalry, TASCAM has wisely given them different missions in life.

Whereas the DA-88 was designed for the professional studio, the DA-38 is aimed at the musician who records at home. The big difference between the two is that the DA-88 features removable printed-circuit boards (PCBs) that slide out from the rear panel, and it has a card slot for such add-ons as the SY-88 sync card. If you're setting up a video-production suite, you'll want to look closely at the DA-88.

But if, as is true of many musicians, your goal is simply to record songs and make CD-quality masters in the convenience of your home studio, the DA-38 is the TASCAM MDM of choice. Not only is it smaller (3U rather than 4U) and more affordable, but according to *Mix* Editor George Petersen, who got a chance to do an A/B comparison of the two machines, the DA-38 sounds better. If you already own one or more DA-88s, you'll be happy to know that the DA-38 will sync to your existing setup as master or slave. The beauty of this arrangement is that the two units have complementary features; locked together, they offer the best of both gene pools.

ALL IN THE FAMILY
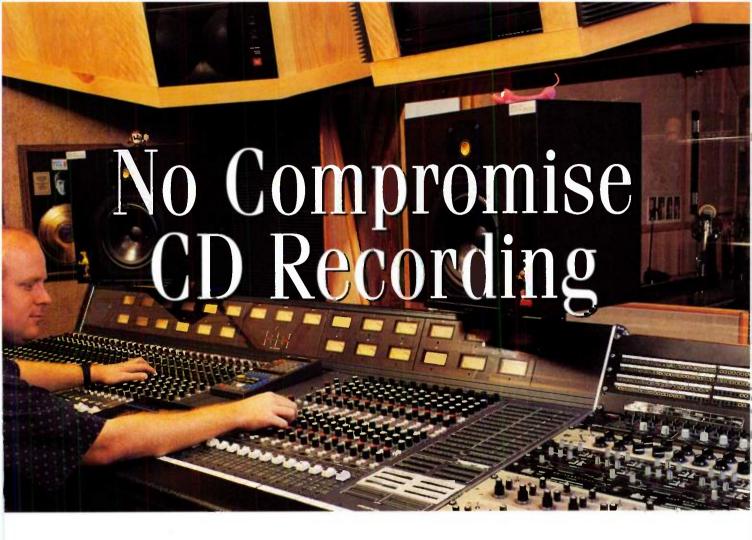
Like the DA-88, the DA-38 uses Hi-8 mm tape to record and play eight tracks of digital audio. Up to sixteen units can be synched for a total of 128 tracks. Like its cousin, the DA-38 employs a rotary, 4-head, helical-scan transport that runs at 16 mm per second and allows 108 minutes of recording time on a 120-minute tape.

Users have a choice of running metal-particle tape (MP), which TASCAM recommends for multitracking applications, or the less-sturdy metal-evaporative tape (ME), which works fine for live and other one-pass recordings. (The manual specifies several brands of each type.) Tape formatting—a requirement with all MDMs—is best done in advance of a recording session but in a pinch can be performed while tracking.

Each new DA-38 comes boxed with a detachable power cord, an owner's manual, a cleaning tape, and a "starter" Hi-8 mm cassette. The RC-808 remote-control unit is optional, but I was sent

TASCAM's affordable new MDM, the DA-38, is tailor made for the home recordist. It offers several unique features, such as an alternate dithering mode and the ability to create composite tracks.

one with the DA-38, so I gave it a go. The RC-808 duplicates all transport controls as well as several other front-panel features and offers track-arming switches for each track. The unit was sturdy, had a wonderfully long cable, and functioned perfectly.

PASSING RESEMBLANCE

The DA-38 sports an uncluttered, well-organized front panel that belies the sophistication and range of its features. A Shift key doubles the duties of the ten function keys. Keypad navigation is a breeze: the word "Shift" and all corresponding functions are silk-screened in blue above the designated keys whereas primary designations are in black. A Menu key controls ten more functions.

The transport-control section has the usual buttons plus three switches: a Format switch that lets you choose between 44.1 and 48 kHz sampling rates, a Digital In switch for recording from digital sources, and a Word In switch for referencing the unit to an external clock source. The DA-38's display window offers easy-to-read numeric display of Absolute (ABS) time and the menu functions plus indicator lights for Format, Track Copy, Digital In, Track Delay, Word In, Machine Offset, the selected sample rate, and Playback (PB) Condition. (The flashing PB light indicates a persistent error in playback.)

The tape port, flanked on either side by recessed power and eject buttons, is located on the upper left side of the unit's face. Eight separate, 12-segment LED-bar-graph level meters (one for each track) appear on the right, with corresponding track-arming switches located directly beneath. A shuttle wheel sits in the middle, below the array of function keys.

The unit's rear panel supplies -10 dBV unbalanced inputs and outputs on RCA jacks, +4 dBm balanced I/O on 25-pin D-sub connectors, TASCAM Digital Interface (TDIF-1) digital I/O, a Remote In/Sync In connector for connecting the RC-848 comprehensive remote controller or for setting up a master/slave sync system using the PW-88S sync cable, a Sync Out connector (also for master/slave setups), a Word Sync In connector, a Remote In connector for the RC-808 basic remote controller, and a Remote Punch In/Out connector for hooking up the optional RC-30P footswitch.

Feuding Features

Feature	DA-38	ADAT XT
Selectable Sample Rates	44.1, 48 kHz	44.1, 48 kHz
Electronic Patch Bay	yes	yes
Track Copying	yes	yes
Composite Track Building	yes	no
Track Advance/Delay	4.17 ms/150 ms	none/170 ms
Recording Time (T120 tape)	108 minutes	40 minutes
Fast-Wind Speed (T120 tape)	1 minute, 20 sec.	2 minute, 20 sec.
Number of Locate Points	2	10
Numeric Keypad	no	yes
Variable Peak Metering	no	yes
Shuttle Wheel	yes	no
Machine ID Numbering	yes: user-assignable	yes: automatic
Adjustable Pre- and Postroll	yes	yes
Crossfade Times	10 to 90 ms (in 10 ms increments)	11 ms, 21 ms, 32 ms, 43 ms
Dithering (for timbral variation)	yes	no
Onboard Oscillator	yes	no
Variable Bit-Length Receptive (digital I/O)	yes: user-selectable	yes: automatic

Thankfully, the RCA input and output connectors are laid out horizontally rather than being stacked tightly on top of one another. RCA connectors may provide better electrical contact than phone connectors, but they're not nearly as user-friendly when it comes to constant repatching. If you do much remote multitrack recording (using the unbalanced I/O), you'll appreciate the horizontal jack layout each time you set up and tear down. To really speed things up, use the D-sub connectors.

FEATURE SHOW

I spent six weeks working with the DA-38, recording numerous singers and instrumentalists (including a full band) and experimenting with its many features. The machine functioned glitch-free and gave stellar results. This unit sounds as good as any MDM I've heard and is simple and pleasurable to operate. I especially love how quiet the transport is. Unlike the DA-88, which has a slightly noisy fan, the DA-38 is all but inaudible during operation (it does not contain a fan), even when you're recording with a supersensitive condenser microphone located only a few feet away. This seemingly trivial detail can actually be critical for the home recordist who does everything—performing, tracking, and mixing—in a single, small room.

The transport is also whippet fast (100 times play speed) and very accurate. The shuttle wheel makes it child's play to dial in the exact location of a punch or song start, down to the exact frame. Some people think a shuttle wheel is superfluous on an MDM, but I find it helpful for many applications. For example, while mixing, there's nothing more frustrating than pushing Play (after the mixdown deck is already recording), only to have the song start from a point that's half a second late or early. With the shuttle wheel, this annoyance is a thing of the past.

One of the musician-friendly features that makes the DA-38 shine is its electronic patch bay. Not only can you assign any input to any track (which means that you can permanently connect, say, a tube mic preamp to an input and then route it to any track you want), you can also use the patch bay in Track Copy mode to bounce sections of tracks internally. This is an awesome feature. For example, you can record several takes of a sax solo and then assemble the best section from each track into an outstanding performance on a separate track. You simply define the beginning and end of each section you want to bounce (while in Rehearsal mode), select the source and destination tracks, and piece together the solo, all in the digital domain. This feature effectively takes the DA-38 a

step in the direction of modular hard-disk recorders.

Another cool feature, track delay, lets you delay individual tracks so they fit into the mix differently. For example, you could delay one track of a doubled vocal so the sound would be a bit "thicker." You also get track advance, which lets you slightly advance one or more tracks. You could use this to give a kick drum a bit more "push," for example. And if you really need to push a track ahead of the rest, simply delay all of the tracks except that one.

Two other helpful features are Vari Speed, which lets you enter a pitch change (±6 percent) for recording or playback, and Repeat, which instructs the machine to play continuously between the two selected Memo points. (This is similar to Rehearsal mode but quicker and easier to get to.) The Repeat feature is especially convenient during mixdown. Rather than be interrupted by continual rewinding, you can remain hunkered over the board, lost in tweakdom, while the DA-38 loops the song indefinitely.

LE MENU

As mentioned previously, the Menu key accesses another ten machine functions. You simply press Shift and then click Menu repeatedly to scroll through the selection. The ABS time disappears from the display window and is replaced by a somewhat cryptic abbreviation for each function. The first menu, for example, is labeled "id SEL 1," which tells you that the machine ID number is 1 (the factory preset). You change the setting by pushing one of the two arrow keys.

Other selections include a menu for setting postroll time, one for setting crossfade times (adjustable from 10 ms to 90 ms in 10 ms increments), and one for switching the shuttle monitor on or off so musicians wearing headphones don't have to hear the tape whizzing by. There's a menu for initializing the backup memory (restoring all factory settings), another labeled "b.E.r." that lets you view the block-error rate displayed on the track meters, and even one for cleaning the DA-38's heads. Pretty nifty.

If you'll be recording from a digital source, you can select a TDIF-input bit length. The unit supports 16-, 20-, or 24-bit TDIF input. I didn't get a chance to use the digital I/O, but there's nothing to it, really: you simply determine the bit length of the source machine and then select the matching number. If recording from another DA-38, for example, select 24; if the source deck is a DA-88, select 16. All that's left is cueing the tapes and pushing Play on one machine and Record on the other. (When recording digital to digital, levels are set internally.)

For those musicians who perpetually

Product Summary

PRODUCT:
DA-38 MDM tape recorder
PRICE:
$3,499
MANUFACTURER:
TASCAM
tel. (213) 726-0303
fax (213) 727-7656
CIRCLE #437 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5				
FEATURES	●	●	●	●	◀
EASE OF USE	●	●	●	●	◀
AUDIO QUALITY	●	●	●	●	◀
VALUE	●	●	●	●	◀

Dueling Specs

Specification	DA-38	ADAT XT
A/D Converter	18-bit Delta-Sigma with 64x oversampling	18-bit Delta-Sigma with 128x oversampling
D/A Converter	20-bit Delta-Sigma with 8x oversampling	20-bit Delta-Sigma with 8x oversampling
Frequency Response	20 Hz–20 kHz	20 Hz–20 kHz
Dynamic Range	92 dB	92 dB
Signal-to-Noise Ratio	92 dB	92 dB
Total Harmonic Distortion	0.008%	0.009%
Channel Crosstalk	90 dB	90 dB

forget to turn their tuners off (battery wasters!), the oscillator menu, which generates a test tone referenced to A-440, may prove a welcome feature. Of course, it's also a handy way to calibrate the meters on your mixer to match those on the DA-38. This is a unique feature on an MDM—or on any recorder for that matter. But perhaps the most unexpected, farthest-out-there feature offered by the DA-38 is the dithering menu. This one even deserves its own subhead.

DITHER THOU GOEST

Dithering is a highly sophisticated process designed to suppress audible artifacts (distortion) that result from digital quantization errors. Basically, dither adds a small amount of noise to an audio signal prior to sampling. This randomizes the distortion to the point of practical elimination while allowing the system to encode amplitudes smaller than the least significant bit. Without dithering, digital recorders have a harder time producing good, clean-sounding audio, especially when recording low-level signals. Therefore, dithering is commonplace on quality digital recorders, no matter what the ilk. The difference here is that TASCAM has added an optional "alternative" Dither mode to the DA-38.

If the unit necessarily uses dithering already, what's the point of having "extra" dithering? I'm not sure what the designers had in mind, but the result is an alternative recording mode that sounds quite different from the normal mode. The DA-38's dithering is operative during recording only. So I recorded a variety of sources—single instruments as well as complete mixes from CDs—at low and high levels and compared the results. On individual instruments, the results were dramatic, but on complete mixes they were less noticeable. Strangely, it didn't seem to matter whether recording levels were hot or wimpy; the tonal difference remained.

Overall, I would characterize the sound as warmer with Dither mode on, with less high-frequency content and a subtle boost in the midrange. This was

● REVIEWS

Price Wars

Item	DA-38	ADAT XT
MDM	$3,499	$3,499
Large Remote Control	$1,599 (RC-848)	$1,499 (BRC)
Small Remote Control	$225 (RC-808); full function	included (LRC); limited function
MIDI Machine Control Interface	$449 (MMC-38)	function of BRC, AI-2 ($1,299), or third-party products
Digital I/O Cable	$110 (PW-88D)	included (fiber-optic)
Sync Cable	$85 (PW-88S)	$15
Footswitch	$35 (RC-30P)	$20
AES/EBU Digital Interface	$1,149 (IF-88AE)	$995 (AI-1)
DASH Digital Interface	$1,399 (IF-88SD)	n/a

especially apparent on solo vocal tracks. The sound of the air around the singer seemed to collapse, reducing the effect of room ambience and making the track a bit less crisp and "live" sounding. In fact, the vocal sounded almost as if it had been recorded through a tube mic preamp and then compressed.

Dither mode nicely tightened the sound of a cowbell, making it sound richer and harmonically more complex, but on a complete mix the difference was so subtle that my ear quickly adjusted to either mode, making it hard to call one preferable to the other. It's incorrect technically, I know, but I found myself thinking of the DA-38's dither menu as being a kind of EQ switch. In effect, the machine offers two different-sounding record modes. Is that a boon for the recording musician? You bet. Variety, after all, is never a hindrance when it comes to recording music.

COMPARISON SHOPPERS

For those of you dying to know how the DA-38 stacks up against the Alesis ADAT XT, we have provided no less than three tables for your comparison-shopping pleasure. That way, you can decide for yourself. (You will note that I've conveniently sidestepped the perennial question of which is the better tape format. From what I've seen, the issue remains unresolved—and judging from the miasma of conflicting opinions and reports, it's likely to remain so.)

I've worked now with both the DA-38 and ADAT XT, and in truth I like them both so much that I'd have a hard time picking one over the other.

Frankly, though, I'd like a machine that combined the best features of each unit. Like the DA-38, it would have a shuttle wheel, a whisper-quiet transport, and the capability of creating composite tracks. Like the XT, it would offer variable track-metering modes, a numeric keypad, and plenty of locate points. Hey, it might even cost less, too!

38 SPECIAL

The TASCAM DA-38 is a rugged, good-featured, great-sounding MDM that would do any home studio proud but can easily hold its own in even the snootiest pro facility. It's fully compatible with the DA-88, and except for not being able to sync to SMPTE or power a meter bridge, it can do practically all the same tricks. And thanks to new-generation converters, the DA-38 sounds better than the DA-88 yet costs significantly less.

The DA-38's particular feature set makes it especially suitable for the home recordist. It even offers MIDI Machine Control when used with TASCAM's MMC-38 interface.

My gripes are barely substantial enough to make a paragraph. However, I wish the DA-38 had at least two more locate points; I can't seem to function comfortably with fewer than four. Also, I would appreciate a peak-hold setting for the track meters. That way, after laying down a track (during which my eyes were closed, of course), I could glance at the machine and determine whether my levels were hot enough or whether, in fact, I had been banging away in the red. Aside from those two minor complaints, I have nothing but praise for the DA-38. ◉

Earthworks OM1 and TC30K

• • • • • • • • • • • • • • • •

By Jim Miller

Capture pristine sound with affordable, precision condenser mics.

When I received matched pairs of the OM1 and TC30K microphones from New Hampshire–based Earthworks, Inc., I was immediately impressed by the custom-crafted, solid–cherry wood cases the matched sets come in. Very classy. Opening the boxes, I beheld four sleek, tapered, brushed stainless-steel instruments that more closely resemble futuristic laser weaponry than microphones. Even before plugging them in, I knew these weren't your average mics.

Earthworks isn't exactly a manufacturing giant that churns out thousands of mics per week. In fact, until recently Earthworks built all their mics by hand, producing no more than twenty units per week. (They've since modernized the production line.)

The company offers three models: the OM1 ($450), TC30K ($500), and TC40K ($900). All are small-diaphragm, omnidirectional condenser mics that require 48-volt phantom power. Many mics cost more for a matched set than for two unmatched mics, but a matched pair of Earthworks mics simply costs twice as much as a single unit. In an Earthworks matched set, the second mic is guaranteed not to vary from the first by more than 0.25 dB at any point across the frequency range.

I tested the OM1 and TC30K, which are virtually the same microphone. These twins exhibit a near-identical, time-coherent, almost ruler-flat frequency response from a remarkable 9 Hz out to 30 kHz. (Time-coherence helps alleviate "smearing" of sounds to deliver better sonic clarity and definition.) The only difference between the two mics is that the OM1 is designed to work with preamps that have electronically balanced inputs (such as those found on boards from Mackie, Yamaha, Soundcraft, and TASCAM), and the TC30K is designed to work with the transformer-coupled inputs found on high-end consoles (e.g., Neve, Summit, and Focusrite). Fortunately, the TC30K also works with electronically balanced inputs, which is nice if you'll be shuttling between pro and home studios.

I ran both models through a Mackie board and detected no sonic or performance differences between them. Obviously, if you know you won't be hooking into a Neve anytime soon, save yourself the extra hundred bucks and go for the OM1.

When using an omnidirectional mic, you often need to use different techniques than with directional microphones. An omnidirectional mic picks up equally in all directions; therefore, there is an inverse relationship between how close it is to a source sound and how much ambient sound it captures. To reduce the amount of room noise it picks up, you simply position the mic closer to the source. Because omni mics exhibit no proximity effect, they can be placed extremely close to a sound source with no increase in bass response.

GUITAR SAMPLES

I was in the middle of several sampling sessions and an album project when the mics arrived, so I was able to give them a vigorous, extended audition. I began by testing the OM1 on a wonderful Taylor 555 acoustic 12-string guitar that I was digitizing for an upcoming guitar-sample CD-ROM. As a test, I began by placing the Earthworks OM1 about a foot away, as I normally would with my trusty AKG C 414 (which I use in the cardioid polar pattern). The results were pleasing but nothing I would rave about.

It was not until I actually moved the mic in close—that is, inches from the edge of the guitar's sound hole—that I began to get a clear picture of what the OM1 could do. Suddenly, the samples came alive with a fullness that literally shocked me. I stopped sampling and just played the Taylor for a while, mesmerized by the beautifully detailed sound that was coming through the headphones.
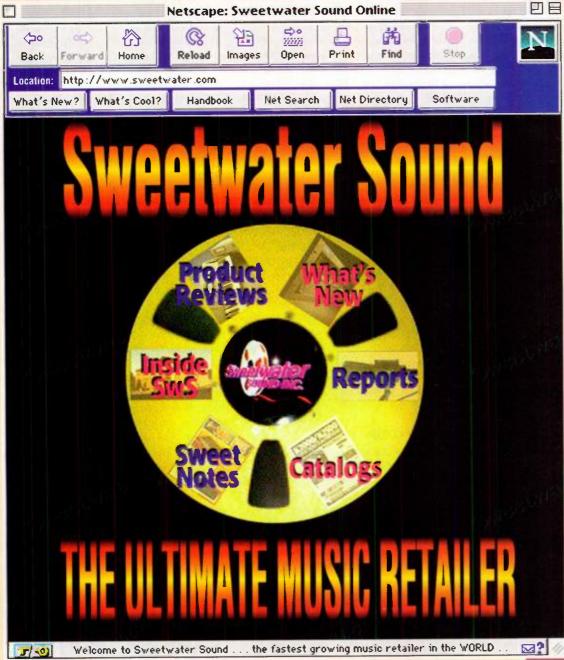
Due to the proximity effect, placing a directional mic that close to the sound hole of an acoustic guitar produces a horrible, boomy sound. But no matter where on the guitar I pointed the OM1s, the results sounded great. I even dragged out my Martin 6-string and gave it a go. (Martin guitars are notorious for their loud body resonance, which is part of why they sound so good when you're in the audience.) Again, the OM1 delivered a wonderfully rich and warm sound, even when placed right up against the guitar body.

Next I plugged a Paul Reed Smith Custom 24 electric guitar into a MESA/Boogie amp, stuck the OM1 right up close to the 12-inch speaker, and let fly with some blues riffs. The sound in

A flat, extended frequency response and ultrasensitivity characterize the line of omnidirectional condenser mics from Earthworks. Shown is a matched pair of TC30Ks. The solid-wood case comes with the set.

my headphones was remarkable. For most of the electric guitars I had sampled up to that point, I used a tried-and-true Shure SM57 mic. The OM1 delivered all the punch I've come to expect from the SM57 but with the added benefit of not being as fussy about mic placement. Usually you need to move a mic around a little bit in front of the speaker to find the sweet spot where the fundamentals and overtones are in balance. But with the OM1, almost all the spots were sweet. And by placing a second OM1 about fifteen feet back and blending the two signals, I achieved a nice, spacious, "live" sound, thanks to the slight time delay between the close mic and the room mic. I later used this combination when adding some guitar leads to an album I was working on.

Next, I plugged a Fender Precision bass into a Peavey amp, shoved the OM1 right up to the speaker, and tried some bright, percussive, slapped tones. Even this failed to daunt the Earthworks mic. The explosive transients were accurately captured, and they eventually ended up on the sample CD-ROM.

BIG EARS
I did discover one possible drawback to these mics. Despite the fact that they can handle incredibly high SPLs (up to 151 dB for the TC30K), they are also incredibly sensitive. I was recording the Taylor acoustic at about 3 A.M., a time when things are pretty quiet in my neighborhood. Although I've taken reasonable care to soundproof my studio, the Earthworks mics captured the sound of traffic out on the main highway a quarter mile from my house. No matter how hard I tried, I could not hear the traffic with the unaided ear. Yet had I turned the volume up on my mixer and listened carefully enough through the OM1, I probably could have told you the name of the song playing on the radio of one of the passing cars.

Earthworks' OM1 is a highly accurate microphone that delivers time-coherent response from 9 Hz to 30 kHz and is inexpensive enough to infiltrate home studios.

Earthworks mentions this very point in their literature. The mics exhibit a 27 dB noise floor and are about 10 dB hotter (more sensitive) than most other condensers. Of course, this could lead to a perception that the mics are noisier than other condensers. For a comparison, I set up my Kurzweil K2000 to play a sine wave through my monitors; plugged in an AKG C 414, a Shure SM57, and the OM1; and set the volume of each mic to produce a +7 dB signal on my board. Then I shut off the K2000 and listened to each microphone through the headphones. As I expected, the OM1 picked up environmental noises that weren't apparent on the other two mics.

This level of sensitivity shouldn't be a problem as long as you are in a soundproof studio or are close miking a loud instrument. Basically, it's a tradeoff: if you want a mic that's sensitive enough to pick up the subtleties of a musical performance, you have to accept that the thing is going to be just as sensitive to sounds you *don't* want to record. From my perspective, the Earthworks mics aren't noisy, they're just ultrasensitive. And of course, they're omnidirectional, so they pick up sounds in all directions.

DRUM IT UP
My next test was miking a drum set for a recording session. The drummer and I miked his kit using two overhead mics, one snare mic, one kick mic, and two OM1s for room ambience. We had little time before everyone showed up for the session, so I quickly set up the OM1s in a fairly close stereo configuration about three feet from the front

of the drums and had the drummer pound out some rock beats. Although the OM1s handled the transients cleanly and presented a spacious stereo image, the drummer and I felt that the sound was not as punchy as we would like. The sound of each drum—including the kick—was accurately captured and finely detailed, but the overall sound was a bit too "distant" sounding for my tastes. (For rock applications, I like an aggressive, in-your-face drum sound.)

So we scrapped the idea of using room mics and moved the OM1s into the overhead position. Suddenly we achieved a wonderfully open sound that had lots of punch and sparkle and included sufficient room ambience without requiring two additional tracks. We ended up using this setup for the rest of the songs on the album.

After the session, I tried a simple experiment. I set up the OM1s in a stereo configuration, moved my guitar amp over next to the drums, and jammed a bit with the drummer while recording only with the OM1s. The result was quite pleasing. I could clearly hear a well-focused sound field with the drums in the center and my guitar coming from the right. Though the sound wasn't as punchy as I like, it was highly detailed and captured what a listener might actually hear if he or she were seated in the room listening to our performance. This leads me to believe that these mics would be ideal for live, stereo recording.

For another percussion test, I positioned a single OM1 close to various individual drums to record single-hit samples. I would normally use an SM57 for this, along with a stereo pair of 414s or a Crown SASS-P stereo mic for the overall stereo mix. The OM1 didn't produce a bad sound by any means, but it lacked a bit of the snappiness and raw energy that I get from the SM57. Probably that's because the OM1 is so accurate whereas the SM57 introduces some brightness in the upper frequencies.

OM1 and TC30K Specifications

Type	condenser
Polar Pattern	omnidirectional
Frequency Response	9 Hz–30 kHz
Maximum SPL (OM1/TC30K)	146 dB/151 dB
Self Noise	27 dB SPL (A weighted)
Weight	8 oz.

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I also tried the OM1 as a kick-drum mic, something I wouldn't ordinarily do with an omnidirectional condenser mic. The OM1 produced a very hefty, room-shaking "whump" that would probably be great for jazz or fusion applications, but it lacked a bit of the "processed" sound I like for rock drum tracks. (I tend to use either an SM57 or an AKG D 112.) Of course, it all depends on the sound you're after. To get the best of both worlds—that is, to produce a kick sound with plenty of snap as well as a deep bass tone—I rec-ommend using two mics, such as an SM57 at the beater position (outside of the drum) and an OM1 inside the drum. If you can't spare the tracks, simply do a live submix of the two tones.

KILLER PIPES

While recording an M. P. Moller pipe organ at a local church for a histori-cal-keyboards CD-ROM, the sensitivity of the Earthworks mics proved prob-lematic. The pipes were located about fifteen feet off the ground, so there was no way to get the mics in close. In-stead, I had to place them about 25 feet away, just in front of the first set of pews. This particular organ was built in 1957, and although it's fairly loud when all the stops are out, it's a rather subdued instrument in its solo voicings.

Unfortunately, I had access to the in-strument only between 9 A.M. and 5 P.M. on a weekday. As you can imagine, I was plagued by environmental sound: traffic, construction noise, a nearby leaf blower, and children at play in the church's daycare facility. Due to the high levels of noise, the Earthworks mics were barely usable. I did record some of the organ's multistop combi-nations to DAT using the OM1s, but for the main sampling work I had to switch to a set of cardioid mics. (Even then some amount of background noise crept onto the tracks.)

However, when I played back the OM1-recorded organ music in my stu-dio, a glorious sound emanated from the speakers, complete with some wall-shaking 32-foot pedal tones that challenged even my subwoofer. The ambience of the church was wonder-fully captured by these mics. I only wish I could have done more of the pipe-organ sampling with the OM1s. Un-fortunately, above even the loudest organ passages, I could readily make out the sound of children screaming and playing in the background. And during the quiet spots, while I was changing stops and making notes, I could actually hear people talking out-side the church as they walked by.

SAX AND VOCALS

I also got great results from the Earth-works mics while I was recording some

Product Summary

PRODUCT:
OM1 and TC30K condenser mics
PRICE:
OM1: $450 ea.
TC30K: $500 ea.
MANUFACTURER:
Earthworks
tel. (603) 654-6427
fax (603) 654-6107
e-mail dblack@jlc.net
CIRCLE #438 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5
AUDIO QUALITY	● ● ● ● ●
VALUE	● ● ● ● ◗

finishing tracks for the album project. For example, I stuck an OM1 right into the bell of a saxophone and another about a foot away. The tone from either mic was really good, but together the two mics produced a killer sax sound. The only problem I ran into was a bit of phase cancellation when the two tracks were summed to mono. This would have been easily enough corrected, though, had I had time during the session to make adjustments.

As we neared the end of the project, we had to reserve one of two remaining tracks for an acoustic guitar (which we also ended up recording with the OM1), so we were forced to put two backup singers on one track. We got a nice recording with the two singers close to a single OM1, but just as an experiment, I had them move back a few feet. This time a bit of distance from the mic proved just the trick for getting the sound we were after. The singers indeed sounded as if they were in the background, but there was still plenty of detail and wonderful clarity.
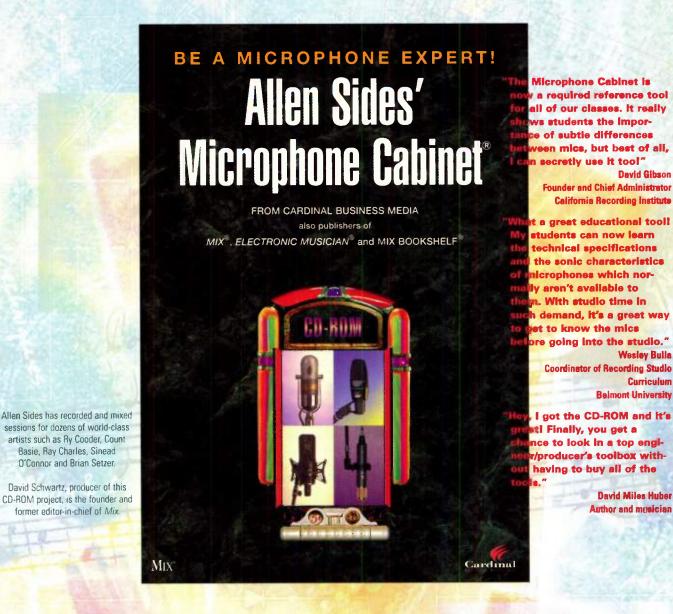
SWEET DEAL
Although the Earthworks OM1 and TC30K are not "do all" mics, either would make an excellent addition to an existing mic collection and would undoubtedly elevate the recording capabilities of most home or project studios. Just keep in mind that they're a different animal from a cardioid dynamic or even from the usual large-diaphragm, multiple-pattern condenser.

These are truly great mics. Used correctly and for the right application, they sound absolutely fantastic, with detail and imaging as good as I've heard. Of course, at a retail price of $400 for the OM1 and $500 for the TC30K, they're hardly in the impulse-buy category. On the other hand, compared to many top-of-the-line, omnidirectional condenser mics (Neumann, B & K, Schoeps, etc.), they're downright affordable. And to give you an idea just how confident Earthworks is about the quality of their product, if you purchase the microphones direct, the company offers a 33-day, money-back guarantee *plus* a two-year guarantee covering parts and labor.

That leaves us with only one problem: the **EM** editors are going to have a tough time getting me to send these mics back! Let's just say they might have to ask more than once.◉

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Wesley Bulla
Coordinator of Recording Studio
Curriculum
Belmont University

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David Miles Huber
Author and musician

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IQS SAW Plus 2.9 (Win)

By Zack Price

A popular digital audio program gets a major overhaul.

t's not easy for software developers to top themselves. Sometimes, the best they can do is add a few missing features that users demand or fine-tune their existing programs so they run bug free. But if software companies are creative and have some foresight, they build on their successes by introducing new versions of applications that rise above the limitations of the original programs.

In the past, Innovative Quality Software (IQS) has demonstrated its ability to fine-tune its programs with monthly (and sometimes weekly) updates. With the introduction of *SAW Plus*, they have shown themselves capable of creating new software that exceeds the confines of their original offerings.

At first glance, *SAW Plus* looks much the same as the original *Software Audio Workshop* (*SAW*, reviewed in the December 1993 **EM**). Although the color scheme is now blue instead of burnt orange, the Track, Soundfile, Region, and Sequence windows are still there (see **Fig. 1**). However, a closer examination of the screen reveals that *SAW Plus* is not just an improved version of *SAW* but a completely redesigned program from the ground up.

SCREENING ROOM

The first thing you'll notice about *SAW Plus* is the high degree of onscreen detail. In fact, IQS suggests that you run the program at a resolution of 1,024 × 768 pixels with 256 colors. The program also operates at 800 × 600 pixels (256-color capability is required in either case), but the higher resolution lets you see more information on the screen at once.

Working at 1,024 × 768 offers no substantial advantage when dealing with lengthy sound files. (You have to scroll the screen at some point!) Still, every bit of extra screen space is always appreciated. Besides, the program's high display resolution is surprisingly easy on the eyes, even when using a 14-inch monitor.

Although *SAW Plus'* default display opens with the window arrangement shown in **Figure 1**, you can customize screen displays to help you perform a variety of tasks more easily. For example, when you're editing a sound file, you might not want to see the Multitrack, Region, or Sequence windows. Furthermore, you want to see as much of the Soundfile window as possible. *SAW Plus* allows you to resize and display selected windows to create custom screens that are geared toward performing specific tasks with a minimal amount of clutter (see **Fig. 2**). You can then store such custom displays and re-call them with different function keys.

Having a set of alternate work screens is handy, but you don't have to design and store a new screen every time you want to temporarily change the display's appearance. By pressing the Page Up key, you can vertically enlarge the active window to see its information more easily. To shrink the window, just press the Page Down key until it returns to its original size.

FOR THE RECORD

To record a track in the original version of *SAW*, you had to call up the Record command from the File menu, name and record the sound file, define and name regions in the sound file, and then place the regions into sequences on different tracks (whew!). Although this hierarchy is the same in *SAW Plus*, the recording process is more flexible and streamlined.

First, you can bypass the region and sequence creation steps by recording directly into one or more of *SAW Plus'* sixteen tracks. Once you name a file (or files, in the case of multiple tracks), *SAW Plus* uses that name to automatically create and name the regions and sequences in that track. This feature is useful when you're using *SAW Plus* as a linear multitrack recorder.

If you know beforehand that you're going to work with different sections of a sound file, you can still record in the Soundfile window. *SAW Plus* allows you to manually place regions into tracks or drag a highlighted region and drop it into the desired track. If the region has not been named, *SAW Plus* gives it one based on the name of the sound file and the order in which the regions were dragged into the Track window.

SAW Plus supports up to four sound cards simultaneously. You can select the sound card through which you wish to route a track's recording and playback in the Multitrack window or the Soundfile window. In either case, you can reroute a track or sound file's output to a different card later.

Depending on the number of stereo sound cards you have installed and on your computer's capabilities, you can

FIG. 1: *SAW Plus'* default main screen, at 800 x 600 resolution, shows the Multitrack, Full View, Soundfile, Sequence, and Regions windows plus the playback transport controls.
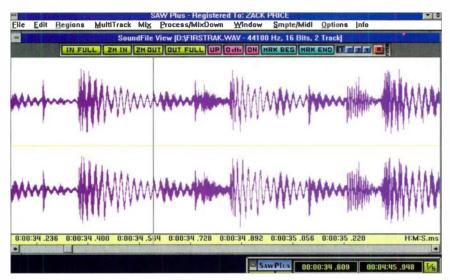
FIG. 2: This alternate work screen was created by selecting only the Soundfile window and playback transport control. The Soundfile window has been resized for maximum visibility.

record up to eight tracks at once in *SAW Plus*. There are several recording options in each card's Record dialog box (see **Fig. 3**). For example, you can record each track as a stereo track, or you can record in Mono L./R mode, where each channel of the stereo sound card becomes a separate monophonic track.

You can also record a monophonic track using Mono L+R mode, which sums the left and right channels into a monophonic track. This mode is particularly useful for making mono tracks using sound cards with stereo miniplug inputs and outputs. For one thing, this eliminates any extraneous noise that can come from recording one channel when the sound card is wired to accept a signal from two channels at once. In addition, the summed input means you

will also have a summed output. This generates an equal signal in both audio channels, which means that you can monitor the recording through headphones without feeling like you're taking a hearing test.

The default input level for each sound card is 0 dB, which is fine for recording sound files individually. However, you may want to turn down the input setting if you're recording multiple tracks at once or simultaneously recording along with existing tracks. Otherwise, you could suffer some serious digital distortion.

In addition, you don't necessarily want each channel of your multitrack recording to peak at a volume level of 100 percent, as shown on the right side of the recording dialog box (see **Fig. 3**). Of course, you can control this with the input-level setting or at the source. In any event, the output levels should usually read about 0 dB, which corresponds to a volume level of about 50 percent. That ought to prevent any nasty encounters with digital distortion crunch.

STAYING ON TRACK

In the original *SAW*, editing was done mainly within the Soundfile window. *SAW Plus* still lets you edit in the Soundfile window, but you now have the

FIG. 3: In addition to the record transport control, two Record dialog boxes are visible in this example because two different sound cards are being used to record on three different tracks.

more elegant option of working directly in the Multitrack window. You can view a track's contents as a waveform (as in the Soundfile window) or as the more familiar solid "tape strip."

Each track includes at least one region, so *SAW Plus* provides a group of Split Region tools to let you edit tracks further. To split a region, move the hairline cursor to the desired split point and then select one of the four Split tools.

The Split/Keep tool splits the region at the cursor point while leaving both parts in place. The Split/Mark tool also splits the region at the cursor while leaving both parts in place, but it also marks the region to the left of the split point. If Select mode is active (more in a moment), the left-hand region is also highlighted. This tool is great for splitting and highlighting many regions for group moving, copying, or deleting later.

The Split/Remove tool splits the region at the cursor and erases the region to the left of the split point. The

Product Summary
PRODUCT:
SAW Plus 2.9 multitrack recording software
PRICE:
$999
SYSTEM REQUREMENTS:
80486 DX2/66 PC (Pentium 90 or better recommended); Windows 3.1 or Windows for Workgroups; 16 MB RAM; SVGA monitor with 800 × 600 resolution and 256 colors (1024 × 768 recommended); Windows-compatible sound card; fast 500 MB or larger hard drive (Fast ATA, EIDE, or SCSI-2)
MANUFACTURER:
Innovative Quality Software
tel. (702) 435-9077
fax (702) 435-9106
e-mail info@iqsoft.com
Web http://www.iqsoft.com
CIRCLE #439 ON READER SERVICE CARD
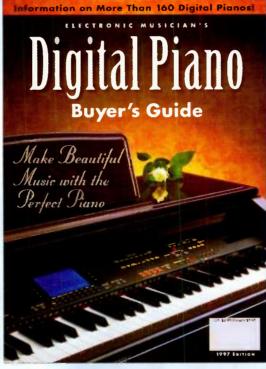
EM METERS	RATING PRODUCTS FROM 1 TO 5				
FEATURES	●	●	●	●	
EASE OF USE	●	●	●	●	
DOCUMENTATION	●	●	●	◖	
VALUE	●	●	●	●	◖

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November 1996

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C. "Multitimbral Muse," p. 78	709	710	711	712
D. "Simply Grand," p. 92	713	714	715	716
E. "Desktop Musician: Creating a Monster," p. 102	717	718	719	720
F. "Working Musician: Executive Ears," p. 124	721	722	723	724

regions to the right of the split point remain in place. Likewise, the Split/Delete tool splits the region at the cursor and erases the region to the left of the split point. However, the regions to the right of the split point are moved to begin at the original starting position of the now-extinct region.

The Cut Multitrack Entry tools offer a more direct method of erasing portions of a track. Just move the cursor to the desired location, and then hold the Shift key and drag the mouse from left to right to highlight the area you wish to edit. You can Cut/Remove the highlighted portion, which erases the section but leaves the regions on either side in their original positions. On the other hand, the Cut/Splice tool removes the highlighted area and closes up the gap by butt-splicing the beginning of the right-hand regions to the end of the left-hand regions.

You don't have to use the Cut Multitrack Entry or Split Region tools to create or adjust regions. You can also drag their end points to resize them. Furthermore, these changes are nondestructive. If you want to expand or contract the region size further, the original data is still there. However, you can't move these points past the start of the track, the boundaries of the sound file in which the region resides, or over other regions.

SMOOTH MOVES

If you activate the Select button in the Multitrack window, you can highlight any regions in any tracks simply by clicking on them or using one of the tools mentioned earlier, which lets you move or copy them *en masse*. The only limitation to moving or copying regions is that you can't place them over existing regions.

For instance, you hold the Shift key and drag the selected regions horizontally within the same track to change their start and end times, and you hold the Control key to drag the selected regions vertically from one track to another. Normally, regions that are moved vertically maintain their original start and end times. However, you can also move regions in any direction by releasing the Shift or Control key once you've started dragging.

Copying regions within or between tracks is much the same as moving them. Begin by using the Shift or Control keys and dragging regions as you

normally would. Once the regions start to move, click on the right mouse button, and *SAW Plus* generates a copy of the regions that you can drag to a new location.

SAW Plus offers a variety of other features that help you move and copy tracks with greater precision. For example, you can use the Marker window (see **Fig. 4**) to set marker points on the fly in the Multitrack or Soundfile window. You can update markers by positioning the cursor at the desired location and pressing the Update button in the Marker window. You can also change the marker name from its default designation to one that's more meaningful to you.

To align regions among different tracks, *SAW Plus* provides a highly useful Snap Region To Cursor function. In addition, you can change the order of the tracks on the screen. For example, say you need to line up a region in Track 2 with the data in Track 7. Just click on Track 7's number and drag it to a track adjacent to Track 2. *SAW Plus* switches the tracks so you can see them side by side. Now, you can perform the edit as you normally would. After you're done, just click on the number of the track you moved, and *SAW Plus* moves the switched tracks back to their original positions.

The most common use for these functions is lining up different tracks to the proper bars and beats. For the most part, they do a good job of making editing sessions easier. However, I wish *SAW Plus* had a bar/beat grid function that could determine measures and major beat divisions from highlighted regions. That would make it easier to edit certain types of material, such as house music, which is often rhythmically precise but without traditional song structures of verse, chorus, and bridge.

POTS AND PANS

SAW Plus now offers nondestructive mix automation with real-time pan and volume control for each track. Just click on the Fader icon for the desired track, and fader and pan controls will pop up (see **Fig. 5**). You can manipulate these controls with the mouse, or for more
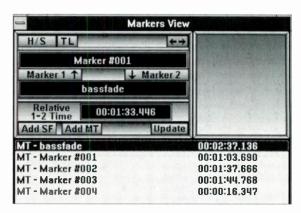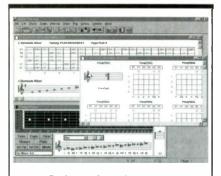
FIG. 4: You can set marker points on the fly from the Markers window. You can also update marker names and locations, shuttle between two assigned marker locations, and jump to selected marker points.

precise control, you can move the cursor to the desired time location and set the volume and pan to the desired values. If you press the Offset button when the cursor is at the start of the track, you can set the amount by which the pan and volume faders will offset subsequent adjustments to these parameters.

Editing volume and pan changes in the Multitrack window is extremely flexible. To move to the next pan and volume change, press the Next button or drag the cursor to the desired location. If you have previously set markers, you can use them to move to the desired editing points. In addition, *SAW Plus* can perform nondestructive linear, logarithmic, and inverted logarithmic volume fades.

If a track's Fader window is open, you can move or copy the pan and volume settings in any highlighted area. For instance, if the area is dragged to a different location within a track, the previous settings are overwritten once the highlighted settings are placed there. You can also move the highlighted pan and volume levels to a different track. The highlighted area's start and end points are retained during movement from one track to another. Moreover, the settings in the original track area return to the default values of 0 dB volume and center pan.

In similar fashion, you can copy pan and volume settings from one track to another. Start by dragging the highlighted area to a different track, and then click the right mouse button while moving to the new track. Again, the start and end points of the highlighted area are retained during movement.

● **SAW PLUS**

However, the original tracks retain their settings.

If necessary, you can clear all volume settings or pan settings within a highlighted area. You can wipe out all volume and pan settings in the entire track by hitting the Clear All button.

PERSONAL EFFECTS

Effects processing in the original *SAW* was only possible with the *SAW Utilities Rack*, which was a $350 option to the main program. All effects processing was destructive and had to be performed offline. *SAW Plus* includes real-time, nondestructive effects processing for little more than the combined cost of the original *SAW* and *SAW Utilities Rack*. Unfortunately, a powerful computer is required to take full advantage of these real-time features.

SAW Plus' most powerful effect is the ParaGraphic Equalizer. Each "graphic" section is a parametric equalizer with adjustable frequencies, bandwidth control over a range of 0.5 to 2 octaves, and 15 dB boost or cut. The Para-Graphic Equalizer also includes adjustable Hi Cut and Low Cut filters (lowpass and highpass filters).

Echo/Delay is probably *SAW Plus*' weakest effect. Delay times for each channel are adjustable in 2 ms increments, which is not precise enough to synchronize delays to tempos. Even though the Echo/Delay features are too basic for my needs, I frequently use the Cross delay parameter, which sends the delayed output of each channel to the opposite channel. When used with short delay settings (under 10 ms), it adds a sense of spaciousness to dry tracks and eliminates that "in one ear" syndrome so common to headphone mixes.

Normalization and Peak Limiting, which are parts of the Comp/Gate/Limiter function, are the only two effects in the entire program that are not performed in real time. To process tracks with these effects, you can destructively process the data in the original files. Better still, you can process the track to a new sound file or an Output Track in the Multitrack window. *SAW Plus* has four Output Tracks that correspond to the sound cards in your system. These Output Tracks are used for monitoring mixes that have been processed using the normal tracks assigned to a particular sound card.

Two effects that might seem totally useless at first glance are the Reverse Phase and Center Channel Eliminator. However, I found them to be lifesavers on more than one occasion. The Reverse Phase effect (which reverses a track's phase polarity) allowed me to achieve better separation and control on a recording with some unavoidable
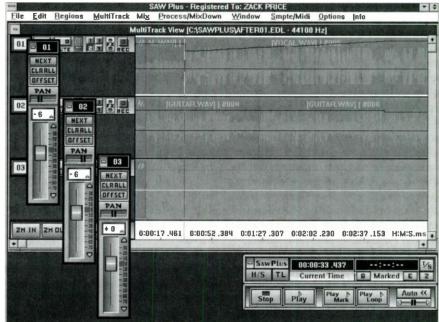
FIG. 5: Pan/Volume faders are available for each track. Pan positions are graphically displayed in the upper half of each track, and volume levels are shown in the lower half. In this example, the diagonal line in Track 1 represents a pan sweep from hard left to hard right.

bleed between the vocal and acoustic guitar mics. Reversing the phase of the guitar track greatly reduced the level of guitar bleedthrough in the vocal track and vice versa.

The Center Channel Eliminator reverses the phase of the left channel and mixes it with the right channel. The most obvious application is removing vocals from a track. However, the success of this process depends on the nature of the original mix. The vocal must be panned dead center or it won't be canceled, and if most of the mix is piled into the middle, there will be little to hear after applying this effect.

This isn't always a bad thing. I used the Center Channel Eliminator to remaster an awful stereo mix that had too much bass, weak vocals and drums, and guitars with no presence. To improve it, I put the stereo mix on Track 1 and heavily compressed the recording using the Compressor/Limiter/Gate. This made the bass more manageable and brought up the vocal and drums to tolerable levels. I copied the original stereo mix to Track 2 and processed it with the Center Channel Eliminator. This silenced most of the mix except for the guitars, which were panned hard left and right. When I played both tracks together, the presence of each guitar was boosted without resorting to potentially harsh EQ adjustments.

SYSTEM REQUIREMENTS
All real-time pan, volume, and effects processing is done without the aid of additional DSP cards or other forms of dedicated hardware. Instead, *SAW Plus* performs all this work with the computer CPU. Therefore, IQS recommends that you run the program on a Pentium 90 or faster with at least 16 MB of RAM and a fast hard drive dedicated to digital audio recording. You can use a slower machine, but program performance will suffer.

If you have a slower machine, a viable alternative is *SAW 6.0*, which is a scaled down version of *SAW Plus*. It has all the real-time features and functions of *SAW Plus* except that it has four tracks and supports only two sound cards simultaneously. Its price is also easier on the wallet ($599 as opposed to $999 for *SAW Plus*).

Interestingly, *SAW Plus* works best on systems running Windows 3.1 or Windows for Workgroups. The program

will run under Windows 95, but IQS doesn't recommend this. Apparently, *SAW Plus* can't access the sound cards' device hooks in Windows 95, which makes playback slightly less stable. In my experience using the program under Windows 95, however, I only heard occasional glitches in playback when I went out of my way to be abusive (e.g., rapidly starting and stopping playback repeatedly).

More interesting was the fact that I experienced no increase in the program's performance under Windows 95, unlike other digital audio software. I suspect this is because *SAW Plus* was designed in 32-bit assembly language, so it already runs at peak efficiency in Windows.

WRAP UP
With its real-time processing capabilities and more streamlined operations, *SAW Plus* is a quantum leap beyond the original *SAW*. Although the effects aren't always dazzling, in most cases they are appropriate for the task at hand, and they help you get the job done quickly and efficiently. Moreover, *SAW Plus* is currently the only Windows program that records eight tracks at once with regular Windows-compatible sound cards.

Aside from the bar/beat grid mentioned earlier, the only other things on my wish list are a decent reverb effect, group fader assignments, and a way to use MIDI Control Change messages to move multiple pans and faders with external MIDI devices. I'm getting tired of interfacing with a mouse.

Right now, the PC digital audio market is exploding, with lots of interesting products competing for your hard-earned dollars. *SAW Plus* more than holds its own against products costing twice as much. Given the nature of IQS's rapid-response approach to software updates and hardware support, I believe that *SAW Plus* will be more resistant to obsolescence than many other programs. Speaking as one who spends way too much time playing with these toys, I have to say that *SAW Plus* is one program that's worth serious consideration. To my mind, it's the best Windows digital audio program you can get for under $1,000.

Zack Price owns and operates Tin Ear Productions, a digital audio editing service in the Chicago area.

Roland A-90

By Steve Oppenheimer

Are you hungry for a great action? This master keyboard delivers.

Good cooking is worth waiting for, or so the saying goes. And when it comes to piano-action master keyboards, Roland-loving electronic musicians were getting pretty famished. After all, Roland hadn't released a new 88-key MIDI controller since the A-80 in 1989, and the genre has advanced a long way since then. It was past time to add some new cuisine to the menu.

But at last, Roland has served up a fresh, hot meal: the A-90. The menu features hearty fare, with plenty of delectable controls and a sweet (but not cloying) user interface. The meal is well presented, and the bouquet is exquisite. But the truth is in the tasting, so please join me at the table, and we'll taste test the A-90. *Bon appétit!*

SETTING THE TABLE

The A-90 features an 88-key, weighted-action keyboard that transmits key Velocity and Channel Aftertouch. The unit stores 64 Performances (user programs) in onboard memory, and you can store another 64 on an optional M-512E RAM card ($129). In addition, you can install one of several expansion boards, which provide a selection of internal sounds.

This is a long, lean, music machine. It's only sixteen inches deep but almost 58 inches long, and it barely fit into the control area of my home studio. It weighs about 60 pounds without the expansion board, which is pretty good for a piano-action keyboard.

Wiring the A-90 is so simple my mother could probably do it, even though she still thinks MIDI is the skirt length between mini and maxi. (But she can cook!) All jacks are clearly labeled both on the rear panel and along the top edge of the front panel. Every electronic keyboard should be labeled in this thoughtful manner.

There are four independent MIDI Out ports, which can be individually disabled with dedicated buttons. Two independent MIDI In ports can be used in several ways, including changing Performances via Program Change, loading settings via SysEx, merging data from an external MIDI controller, and playing the internal sounds from a sequencer or external controller.

The In ports offer some handy bells and whistles. Data appearing at MIDI In 2 is simply merged with A-90 data in the traditional manner and routed to any Out port without being processed. However, MIDI In 1 is labeled "Remote" because it lets you process and route channel messages from a remote MIDI device (say, a MIDI guitar) through the A-90 Zones. The A-90 receives the data in Omni mode; the Zone parameters determine the output channels and ports. I especially appreciate the ability to disable the A-90 keyboard for any Zone so that only the external device controls the Zone. That way, the A-90 and a sequencer or second controller can independently access any internal sound or connected MIDI device while applying the Zone parameters.

The L/R audio outputs for a sound-expanded instrument are on ¼-inch TRS jacks that can be used with balanced or unbalanced (TS) lines. You also get a stereo headphone jack. A rear-panel slot admits Roland M-512E memory cards, which can store system settings, Manual mode settings, up to 64 Performances, and chain data (which I'll explain shortly).

Other rear-panel features include six ¼-inch controller inputs, a breath-controller input, a power switch, and a permanent, 2-conductor power cable. I prefer a 3-conductor power cable for optimal grounding, but in practice, I had no problems with ground loops.

EATING WITH YOUR HANDS

Keyboard actions are difficult to discuss objectively unless you get into the minutiae of their construction. The critical aspect is subjective feel, so you should play the A-90 at your local dealer before making a decision.

That said, I love Roland's proprietary PA-4 hammer-action, cantilever-based keyboard. The action isn't stiff, but it is heavy enough to make this veteran pianist feel at home. (I have a fairly light touch, but I abhor feathery-light actions.) I was able to smoothly transition through a full range of dynamic variation without stress. Even fast trills and tinkly high-end parts played with a very light touch came out clean and even. Thanks to Roland, we have an excellent alternative to the various Fatar-built weighted actions that are employed by most master-keyboard manufacturers.

ENTICING PRESENTATION
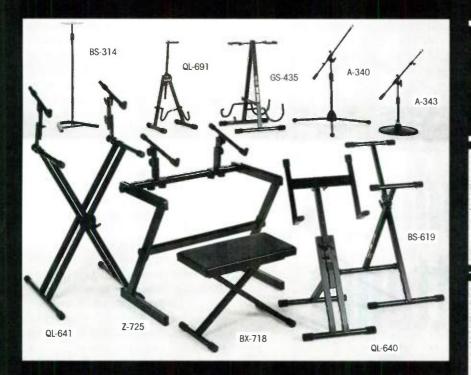
I was immediately attracted to the A-90's logically organized control surface, though I needed some help from the manual to grasp the unit's operational modes. As I'll detail shortly, the A-90 lets you define up to eight independent key Zones that can address external MIDI devices or the optional internal sounds. Setting the basic parameters for each Zone (Key Range, Volume, Pan, etc.) is a no brainer. These parameters are easily accessed with dedicated buttons.

Some controller-assignment, global, and utility functions are buried beneath a few levels of software pages. These parts of the user interface are a mite less accessible than I'd like, mostly because the main LCD (there are two) is small and uses abbreviations and symbols. Nevertheless, I figured

The Roland A-90 MIDI master keyboard offers a smooth 88-key piano action, a mostly friendly user interface, and a variety of programmable controllers. You can add sounds with your choice of three optional voice-expansion boards.

out these features quickly enough with occasional help from the manual.

The A-90's buttons have a mottled surface of raised dots, which provides tactile feedback and superior grip. All buttons include inset status LEDs except the Song Select and Reset buttons in the sequencer section and the buttons in the data-entry section. This visual feedback allows you to ascertain the status of your most important settings at a glance, even in poor light conditions. I'm a sucker for this type of design; I knew right from the be-

ginning that the A-90 and I would get along fine.

The front panel is divided into functional sections and subsections, with the two LCD screens forming the focal point in the lower center. Above the dual LCDs and dominating the central portion of the front panel is a large plastic plate that protects the expansion slot. Stenciled on the plate is a partial flow diagram and a short list of the hidden software pages. This list of pages is nice; the rest of the information on the plate is superfluous. I wish

Roland had used this central front-panel real estate for a larger display. But aside from the cost of the larger display (which is worth paying, in my estimation), that would necessitate re-locating the expansion slot, which does not appear to be a simple proposition.

The real-time controllers—two volume sliders, four Palette sliders, three buttons, two wheels, and a joystick—are on the far left, with the wheels on the upper control surface, not adjacent to the keyboard. Between these controllers and the dual LCDs is the main programming area, which includes sixteen Parameter Select buttons, eight Zone-select buttons, and four buttons that enable the MIDI Out ports.

The data-entry section to the right of the display contains a slider, four cursor keys, increment and decrement buttons, an Enter button, and a 10-key alphanumeric pad. An Exit button gets you out of certain programming pages, and an Off button disables the selected parameter. Two other buttons (labeled "A/INT+" and "B/CARD-") do multiple duty, switching between positive and negative values for appropriate parameters (such as Pan and Transpose), toggling between internal memory and a data card, and selecting uppercase letters when you name objects.
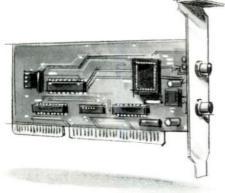
Unlike those on many Roland devices, the A-90's Inc/Dec buttons only scroll through values at low speed and do not accelerate when you continue to hold them. However, the data-entry slider can get you around pretty quickly, and you can adjust the values of most Zone-specific parameters with the four Palette sliders.

Roland also gave the A-90 a rare feature for a MIDI keyboard: a Help button. It's an interesting idea, but in most cases, the information provided by this feature is already made obvious by the main display and status LEDs. The Help function does give you the available range for the selected parameter, which is nice. But when you get to the deeper features, for which help might really be needed, the A-90's Help fails to provide meaningful enlightenment about how the functions work and what they do. Then again, it's hard to provide meaningful help on a 2-line × 17-character display.

To the right of the data-entry section are the sequencer-control, panic, and Function (mode) buttons. Five more buttons are dedicated to controlling

an external sequencer. These include Song Select L/R, a Tempo button that takes you directly to the Tempo page from any mode, and a single Stop/Start button. When you press Start, the A-90 actually sends a Continue message, so your sequencer begins playback wherever it left off. In lieu of a true Start button, you get a Reset button, which sends a Song Position Reset message that stops playback and sends the sequencer to the top of the song.

The A-90 constantly sends MIDI Clock and Active Sensing. Unfortunately, these can't be defeated, which forces you to filter them out if you need to reduce the data density.

WHEN INDIGESTION STRIKES

One of the most common hazards of MIDI devices is the hung note, which usually occurs when, for some reason, a sound module responds to a Note On but not to the corresponding Note Off. To keep hung notes from sustaining *ad infinitum*, every master keyboard should include a panic button that tells your devices to chill out. The A-90 delivers a fine implementation of this crucial feature.

The simplest type of panic button just sends All Notes Off messages on all channels. But some synths, mostly older ones, don't recognize this message. The A-90 doesn't even bother with All Notes Off; instead, pressing the panic button sends individual Note Off and Sustain Off messages for all notes and Zones *that are currently playing*. The A-90EX also resets its internal sounds. If you just want to reset the internal sounds without sending the other messages, you simply hit the panic button without playing any keys on the keyboard. The button's LED flashes while this is going on.

If you double-click the panic button, the unit sends Sustain Off and Note Off messages for all MIDI notes (0 to 127), *whether they are currently playing or not*, along with Volume = 127, Pitch Bend = center, Channel Aftertouch = 0, and Modulation = 0 on all MIDI channels. Again, the EX version also resets its internal sounds. The MIDI In and all other operations are temporarily disabled until all these messages are sent. Now that's what I call a panic button!

SETTINGS A LA MODE
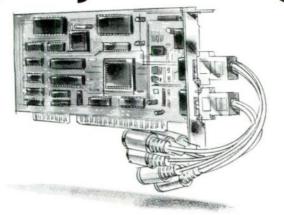
The A-90 offers two types of settings: System and Performance. System set-

tings are global and do not change when you switch Performances. A Performance is the A-90 equivalent of a synth program and includes all the device-specific control values you would expect to change when you switch sounds or effects.

The unit operates in one of five different modes. When you fire up the A-90, it is in Performance mode, which is the mode most often used for live performance. In this mode, Performances can be recalled via Program Change messages, settings can be changed in

real time with the Parameter Select buttons, and the sequencer controls are active. Any Zone-specific settings you change in Performance mode are lost when you change Performances, unless you've saved them as a new Performance. (System settings, on the other hand, are always saved automatically.) Unlike the other modes, there is no dedicated button for Performance mode. You enter it simply by exiting any other mode. Thus, the A-90 has only four Function buttons for five modes.

Manual mode is primarily intended for studio applications. You can't switch Performances in this mode, which makes sense; in the studio, you rarely have to change all your settings at once on the fly. Instead, you program a Performance for a particular sequenced song and make changes directly to that Performance or in your sequencer. The sequencer controls are still active, of course, and you still have real-time control in the Parameter Select section. But unlike Performance mode, *all* Manual mode parameter changes—not just system settings—are automatically saved. Roland made an excellent design decision here.

Chain mode is strictly for live work and has few distinguishing features other than letting you create and step through a chain of any 64 Performances. You can save up to ten of these chains. The sequencer buttons and real-time controllers (Palette sliders, wheels, etc.) are active, but the Parameter Select buttons are not. The idea is to let you step quickly through the Performances required for a song or medley while making limited on-the-fly changes with the controllers.

Edit mode allows the most in-depth parameter editing. You can get at the system settings, globally assign the various controllers to their functions, name Performances, and edit MIDI In and MIDI Out assignments for each Performance. All parameters can be reached via numeric-keypad shortcuts or the programming buttons. Controller assignments are a breeze in Edit mode: when you move a real-time controller, the LCD jumps to the appropriate assignment page.

Utility mode offers special functions, such as saving Performances, dumping and loading SysEx, copying parameter values between Performances, accessing memory cards, adjusting the Voice Expansion board's Master Tune setting, and reinitializing the unit.

Finally, there is a huge variety of system settings. In addition to accessing the expected global parameters, you can customize several aspects of the user interface, including the LCD contrast, whether the unit powers up to default Performance 1 or to the last active Performance, whether Bank numbers are displayed with Performance numbers, and much more.

THE EIGHT-COURSE MEAL

The A-90 provides up to eight independent, user-namable Zones with programmable ranges, which can overlap. Four Zones control the internal sounds and four address external MIDI devices. As a result, the Zones are labeled Internal A, External A, Internal B, External B, etc.

Each of the eight Zones can be activated with a dedicated button, and you can play with any combination of Zones. You can solo a Zone by double-clicking on its activation button. In addition, you can program any internal Zone to address any MIDI Out port, giving you up to eight external Zones, even if you have the Voice Expansion board installed. This is the height of flexibility. However, you can't simultaneously edit both Internal and External Zones; you must edit one type at a time.

Two Parameter Select buttons, labeled Aux 1 and Aux 2, are programmable switches that can send a huge variety of messages to External Zones.

SONIC SPICE

I tested the A-90EX, which is an A-90 that comes with a preinstalled VE-RD1 sound-expansion board. The VE-RD1 provides 128 preset sounds, of which more than half are pianos, electric pianos, clavinets, and organs. You also get a small assortment of string sections and synth strings, a smattering of synth brass, a good variety of basses (but no electric fretless), some synth leads, and a bunch of pads. But you won't find regular brass, solo strings, woodwinds, guitars, drums, or ethnic instruments. Except for the pianos, the VE-RD1 seems mostly designed for playing pop.

The pianos are the main attraction, featuring a variety of mellow and bright timbres with varying degrees of attack. I like the stereo grands best; I used two for a rock session and found they supported other parts well, yet they cut through the mix when needed. They held up well for solo jazz (post-bop), too. These are good pianos that can fill many roles. The stereo semigrands and the full grands are good, though the very bright ones are

too much for me. Some darker pianos sound too obviously like filtered (read: unconvincingly dull) versions of the others. But overall, Roland gets high marks here.

The essence of the Rhodes and Wurlitzer sounds are well reproduced in the RD1, but they're awfully clean. A genuine Rhodes breaks up into distortion when you really smack it; the VE-RD1 is far too polite. For the average listener, these patches will be fun to play, but they won't hit diehard Rhodes lovers where they live. The Wurlitzer seems closer to the mark, though I admit that I haven't played a real Wurli recently. Most of the electronic pianos and pads are pleasant, but you can get their equivalent almost anywhere.

The VE-RD1 has excellent bottom end, which helps fill out the organ and synth sounds a bunch. Of the four B-3 samples, "B-3 Organ 1" is my favorite. It has plenty of gusto and sits nicely in a rock mix. It sounds even better if you add a little bit of analog "fur" with a tube preamp. The mod

wheel controls the depth of a weak Leslie simulation that "rotates" at a preset speed.

The remaining sounds range from pretty good to so-so, with no big winners and few outright losers. The acoustic string sections are suitable for ballads, and the "Dyna Marimba" attacks with a nice woody "thump." I'm not really a clavinet player, but the sample sounds pretty good. You get nine basses, all of which display decent quality.

When it comes to synth strings and synth brass, any good polyphonic, analog synth is fatter and offers more timbral change over time than you'll hear from wavetable synths of this caliber. That said, the VE-RD1's strong low end helps the synth brass, in particular. As to the RD1's synth strings, well, you can have my share of them. The rest of the synth sounds are mostly pads of the type that you love for one particular pop song and then never use again. You'll find a bank of such sounds on every wavetable synthesizer.

In addition to sending CC 0 to 119, Aftertouch, and Pitch Bend (with fixed values), you can use these buttons to set Pitch Bend Sensitivity, Coarse and Fine Tuning, settings made with MSB or LSB (such as Bank Select), and the various GS filter, envelope, and vibrato parameters. You can even program and send SysEx; when you hit the Aux button, the assigned message is sent.

The A-90 simultaneously displays selected values for all four Zones, which is a big advantage for such common needs as balancing levels and setting relative pan positions across Zones. It would be even better if you could see and edit the values of all eight Zones, but no dice. Oh well, the LCD isn't large enough to show eight Zones anyway (sigh).

Any Zone can be routed to any of the four MIDI Out ports. Not only can you independently activate each Zone, you also can independently activate each MIDI Out. As with Zones, you can double-click any MIDI Out button to solo that port. In fact, you can pretty much turn off or on whatever you want whenever you need to.

THE MAIN COURSE
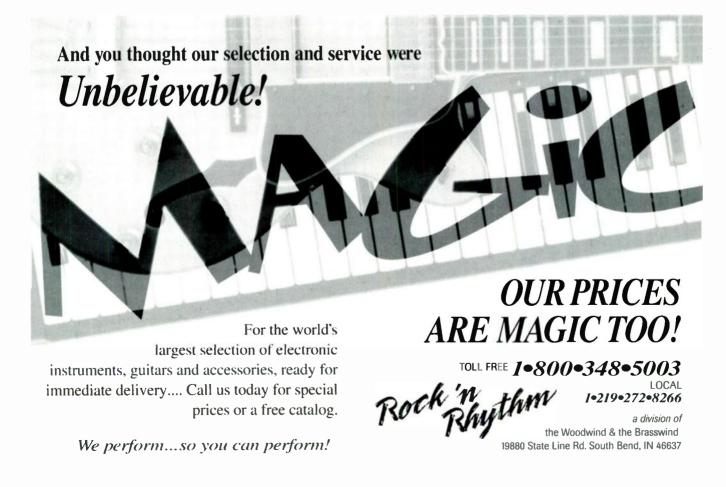
One of the strongest features of the A-90 is its bevy of sixteen programmable MIDI controllers (not counting Velocity, which is part of the MIDI Note On message). To begin with, you get the keyboard's Channel Aftertouch and two traditional control wheels: wheel 1 is spring-loaded and defaults to Pitch Bend, and wheel 2 defaults to Modulation. Below the wheels is a joystick that defaults to X axis = Pitch Bend and Y axis = Modulation. As for key Velocity, you can set the Velocity sensitivity and select one of seven Velocity curves for each Zone. The LCD graphically displays the curves, which is neat.

There are two volume sliders: Internal and Total. The latter controls the combined levels of all internal and external sounds. The A-90 has no master volume control for the External Zones, as such. However, you can disable the Total Volume slider for each Zone, so by defeating it in the Internal Zones, you can make it into an External master volume control. The two Volume sliders are adjacent, so you can easily grab them both to fade everything.

Four programmable Palette sliders are also available. When editing a Performance, each slider controls the active parameter for one Internal/External pair of Zones (e.g., slider A controls the parameter values for both Internal A and External A). Using one slider for both Internal and External control makes a lot of sense for editing, given that you can't simultaneously edit Internal and External parameters anyway. Even in Performance mode, where all eight Zones can be active, four sliders will be sufficient for most people. Remember, you also have a dozen other controllers.

Jacks are provided for two programmable footpedals, two programmable footswitches, a hold (sustain) pedal, and a volume pedal that duplicates the function of the Total Volume slider. (A Roland DP-6 piano-style sustain pedal is included; the other pedals are not.) Musicians with oral fixations will love the breath-controller input, which you don't find on most master keyboards.

Each controller can be assigned to send a long list of messages, including

● A-90

A-90 Specifications

Keyboard	Roland PA-4 hammer action
Velocity/Aftertouch-Sensitive	Yes/Yes (Channel)
Zones	4 Internal*, 4 External
Name Maps	4 user, 9 preset
Performances	64 internal, 64 opt. memory card
Chains	10
Display	2 backlit LCDs: 1-line x 3-character, 2-line x 17-character
Foot Controller Jacks	2
Footswitch Jacks	2
Other Controller Jacks	Hold, Total Volume, Breath Controller
Audio Outputs (for expansion cards)	¼-inch TRS
Dimensions	57⅓ (W) x 16 (D) x 5⅝ (H) inches
Weight	60 lb. 4 oz.

*Can be configured as External Zones.

Control Changes 0 to 119, Channel Aftertouch, Poly Aftertouch, and Pitch Bend. The A-90's Poly Aftertouch can affect the highest, lowest, first, or last note played. True, this isn't as good as having a keyboard with real Poly Aftertouch, but no 88-key weighted keyboard sends Poly Aftertouch. I'm pleased the A-90 supports this control message in some form.

The switch-type controllers can send MIDI Mode messages (All Notes Off, All Sound Off, Reset All Controllers, Local Control, Omni On/Off, and Mono/Poly), or they can increment through Tempo values for a sequencer or drum machine. You can also step through A-90 Performances in numerical order or step through a programmable chain of any 64 Performances. Finally, you can incrementally lower (but not raise) MIDI Volume values from a footswitch. I haven't figured out a practical use for this, but it's there.

Below the Palette sliders are three global control buttons: Transpose, Mono, and Portamento. Transpose lets you globally transpose by ±36 semitones. (You can transpose individual Zones using a Parameter Select switch.) By default, Mono toggles between monophonic and polyphonic modes, and Portamento switches the "glide" function on and off. However, these two switches can be programmed for other functions, such as sending Control Change messages.

By default, the A-90 sends Reset All Controller messages to each Zone whenever you change Performances.

However, this can sometimes be a disadvantage, especially with the sliders because the sliders' actual values are not necessarily the same as their apparent values. For instance, let's say you use a slider to bring the reverb all the way down on Zone A and then you change programs. If the unit sends a Reset message, the next time you move the slider, you'll be starting with the Zone's new default value (whatever that may be), even though the slider appears to be set to zero. This was one of the biggest complaints about the old Roland A-80 master keyboard.

Thankfully, the A-90 lets you change the default system setting so that Reset All Controllers is not sent when you change Performances. That way, the last value sent by a slider will still be in effect for the same Zone in the new Performance. As in the previous example, you can lower the reverb to zero and change Performances. When you next move the slider, the reverb value will start at zero, which was the last value you sent. You also can invert any fader so that the values decrease as the slider is pushed up. For example, this lets you create crossfades.

Despite all this power, the A-90's controller implementation is not as complete as that of some other master keyboards, such as the Peavey C8, Kurzweil PC88, or Fatar 2001. For example, the controller curves are only applied to Velocity, you can't program your own curves, and you can't program offsets that start the Velocity curves at values other than zero.

In addition, although you can disable controllers and set their values independently for each Zone in each Performance, the controllers' functions (i.e., what parameter a given controller affects) are global settings that can only be changed in Edit mode. According to Roland, this was an intentional limitation to simplify programming. That's understandable, but it means that you can't use the same controller for different purposes in different Zones, which would allow you to control complementary parameters in layered sounds or change reverb time and level simultaneously onstage. You might be able to work around this limitation by editing your sounds to create the appropriate maps, but that's a lot more work, and it assumes the modulation maps in your sound sources can be edited.

SPICES AND CONDIMENTS
The stock A-90 has no onboard sounds, but you can easily add your choice of sonic spices; like many recent Roland synths, the A-90 can accept one user-installable VE-series voice-expansion

▼

Wiring the A-90 is so simple my mother could do it.

board. Currently, Roland offers the VE-RD1 64-voice piano expansion board ($545); the VE-JV1, a 28-voice board that essentially contains a JV-90 sound engine ($475); and the VE-GS1 28-voice GS (Sound Canvas–style) expansion board ($395). I tested the A-90EX, which is an A-90 with a preinstalled VE-RD1 board (see the sidebar, "Sonic Spice").

The 4-part multitimbral VE-RD1 was designed for the A-90, and all its parameters can be accessed directly from the Parameter Select section. However, the Parameter Select buttons don't access the parameters on the VE-JV1 or VE-GS1. In addition, although the VE-JV1 is 8-part multitimbral (including the rhythm part) and the VE-GS1 is 16-part multitimbral, the A-90 can only address four Parts because it only has four Internal Zones.

The sound-editing features for the VE-RD1 are relatively basic. You get an

● **A-90**

ADR amplitude envelope, a Bright parameter that opens the filter, and control over volume, pan, fine-tuning, and transposition. Other than the onboard effects, that's it.

The onboard effects processor delivers reverb, chorus, and 3-band EQ. The effects include a solid set of parameters, including reverb type, level, time, high-frequency damping, and feedback as well as chorus level, rate, depth, predelay, and feedback. You can independently adjust the reverb and chorus send levels and configure the effect outputs in parallel or serial. The EQ has fixed-frequency high and low shelving bands and a sweepable mid frequency. All effects parameters except the output configurations can be MIDI-controlled in real time. This degree of control is great for an add-on sound board. I can think of numerous outboard effects processors with fewer editable parameters.

The 128 expansion-board patches can be assigned to any of the A-90's Performances. Most of the preprogrammed Performances use both Internal and External Zones, so the results depend on what synths you have attached. I couldn't use any of the Performances out of the box without editing or disabling the External Zones.

I think expandability is a great concept, and the VE-RD1 has some fine sounds, especially the pianos. But if I can only choose one expander board with full parameter support, I would prefer a wider selection of sounds that includes all the major food groups, as with a GS set. Most users will need to

Product Summary
PRODUCT:
A-90 MIDI master keyboard
PRICE:
A-90: $2,450
A-90EX: $2,995
MANUFACTURER:
Roland Corporation US
tel. (213) 685-5141
fax (213) 722-0911
Web http://www
.rolandus.com
CIRCLE #440 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5			
FEATURES	●	●	●	●
EASE OF USE	●	●	●	●
DOCUMENTATION	●	●	●	●
VALUE	●	●	●	◖

use an external synth with the A-90EX. That's not a major tragedy; after all, a master keyboard is designed to control external devices. But I'd rather pay a few more bucks for a one-stop gigging machine.

SIDE DISHES
To my knowledge, no other MIDI master keyboard offers a feature comparable to the A-90 Name Maps, which let you recall programs from your outboard gear by name, rather than by number. You can create up to four custom Name Maps (which are saved automatically as system settings) with this excellent feature, and preprogrammed Name Maps are provided for several popular Roland synths, such as the JV-80, JV-90, JV-1080 (including the SR-JV80 expansion board), JD-900, SC-55, SC-88, P-55, and M series.

Name Maps are an especially great boon for live performers, but I also used this feature during rehearsal and for a session in which I tracked directly to ADAT. (Normally, I would use a software sequencer that calls programs by name by integrating with a patch librarian or via custom program lists.) Name Maps are also handy if you use a hardware sequencer (say, on a keyboard workstation). I like Name Maps, but think how much cooler they would be if the A-90 polled your external devices via SysEx and created Name Maps automatically, sort of the way that Opcode's OMS does. Hey, it's just a suggestion.

DESSERT
Roland has done a fine job with the A-90. I enjoyed playing the keyboard, and I certainly can't complain about a lack of controllers. It's extremely easy to access most of the important parameters, and I love the wealth of buttons that have status LEDs. When I used the A-90EX at a friend's project studio, I got through the day-long session without referring to the 2-volume documentation except for the patch list.

Back in my home studio, when I referred to the manuals regarding the deeper features, I quickly found what I wanted. The documentation is not perfect, but it's quite good, especially by Roland standards. (I wish Roland would stop calling outboard effects processors "effectors," though; I initially thought the term referred to a special feature.)

I do have some wish-list items. I've already mentioned most of them, so I'll just recap a few of the main ones. For starters, the A-90 doesn't allow each real-time controller to send different messages for each Zone, nor does it let you apply curves to any controller, create controller curves, or apply offsets. I don't need all of that, but most of it would be helpful. In addition, the A-90 has some hidden features, and its small LCD requires abbreviations, both of which detract from an otherwise friendly user interface. And I really don't like the fact that the internal sounds are muted when you change Performances.

The A-90's list price is not out of line, but it is at least $150 higher than several equally well-endowed competitors (e.g., the Fatar 2001, Peavey C8, and Kurzweil PC88) that I have reviewed in past issues of **EM**. However, in one noteworthy case, the price gap is much greater: the Peavey C8p lists for just $1,599. Of course, there are advantages and disadvantages to each unit, but overall, the A-90 does not deliver significantly more features for the extra bucks. Then again, all of its competitors use Fatar actions; if you prefer the Roland action, the other differences may not matter.

A big plus is the fact that Roland has made very good keyboards and excellent sounds for many years, and the company stands squarely behind its products, which is worth some extra dollars. Furthermore, of the dedicated 88-key master controllers (as opposed to synths with 88-key actions but fewer control features), only the A-90EX and PC88 have onboard sounds.

Finally, don't underestimate the fact that you can get to the main programming parameters on the A-90 much faster than you can with most master keyboards and you can see what's going on at a glance. When the studio clock is running or you're making last-minute edits at a gig, that instant access can be worth a whole fistful of extra dollars.

In short, Roland has served up a delicious meal. As with going to a top-flight restaurant, if you like the menu, it may be worth digging a little deeper to pay for it—or even washing a few dishes.

EM Senior Editor **Steve O** *has reviewed every major 88-key MIDI master keyboard on the U.S. market since 1990.*

Intelligent Devices Pro Audio Analyzer (Mac)

By Paul D. Lehrman

Find out things you never knew about your recordings.

Computers make wonderful tools for music production. As recorders, editors, MIDI and studio controllers, signal processors, and sound-designing systems, they have insinuated themselves into almost every aspect of audio. One area in which they have lagged behind, however, has been analysis of recorded sound. To be sure, many digital audio programs allow visualizations of waveforms, and some (such as Steinberg's *WaveLab* and Digidesign's *Sound Designer*) have spectrum and/or 3-dimensional Fast Fourier Transform displays. But these are static displays that can look at only an instantaneous slice of sound.

True real-time audio analysis, usually handled by a rack of pricey dedicated hardware, involves multimode level-metering, instantaneous displays of the relative levels of different frequency bands within the audio spectrum, and 2-dimensional displays of the relative phase of stereo signals. Intelligent Devices' *Pro Audio Analyzer (PAA)* provides all this for the first time on a Macintosh computer, using the Mac's built-in audio hardware or Digidesign's audio cards. The software applies all of these analysis tools and more to mono or stereo audio files using fast, multicolored, easy-to-understand displays.

VARIATIONS ON A THEME

Pro Audio Analyzer comes in two versions. The stand-alone version is for use with external audio sources, which can be routed into a Power Mac through the computer's internal CD player; its 16-bit, 44.1 kHz audio inputs; Digidesign Audiomedia II NuBus card; or Digidesign Pro Tools I or II system. (Pro Tools III, which includes TDM, is not supported by the stand-alone version.)

The TDM version is a plug-in for Pro Tools II or III systems. It uses one DSP chip on a DSP Farm card. The plug-in version of *PAA* shows up in the host program as an insert, either in mono on a channel or in stereo on an auxiliary input.

You can hear the audio that is passing through *PAA,* and several monitoring options are available in a dialog box (accessed with Option-clicks) in the plug-in version or from a menu in the stand-alone version. You can monitor an incoming stereo signal, summed mono (L + R) on both channels, the difference signal (L - R) on both channels, or the difference signal on the left channel and the summed signal on the right. The output can operate at full level, or you can apply a 20 dB pad. An output mute switch is also included.

LEVEL METERS
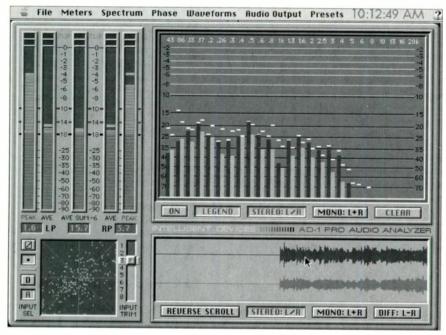
The program's window is divided into four parts, which can be viewed together or in various subsets. The leftmost section shows levels for each stereo channel, with independent peak and average (VU) indicators. Each channel uses the conventional green ("safe") and yellow ("warning") zones, topped with a red clip light. These meters can be set to look like LEDs (large segments), plasma displays (small segments), or continuous displays. The meters cover a range of 96 dB, which means they're almost always showing *something*—input noise, converter noise, hum, and so on—unless you are playing a signal that consists of mathematical (digital) silence.

Above the colored segments are small, white bars for peak-hold indication. You can independently adjust the meters' release rate and the peak-hold indicators' decay rate, with a choice of four speeds.

A fifth level meter in the center of this section can show the average or peak levels of either the sum or the difference of the two channels (a 6 dB pad keeps the summed signal close to the same level as the individual channel meters). A trim control adjusts an analog input signal in eight steps over about a 20 dB range. (This control has no effect on digital signals.) The steps are uncalibrated, but the zero position is clearly marked.
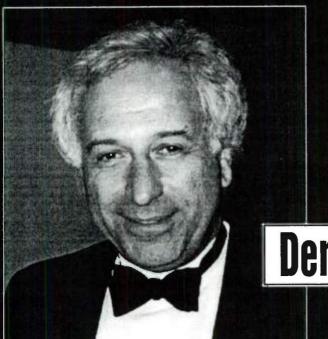
The two clip lights atop each meter can be set to trigger at anywhere from 1 dB to 0.05 dB below the maximum digital audio level. With resolution that fine, you can push your levels really close to the max if you want to be daring. One clip light goes on at the first sign of clipping and stays on until you reset it, and the other resets itself after a while. Below the meters are numeric peak indicators that show, to the nearest 0.1 dB, the highest level encountered.

The manual makes a strong point of saying that *Pro Audio Analyzer* is far more accurate and reliable than any other digital metering system because it

Intelligent Devices' *AD-1 Pro Audio Analyzer* **for the Mac provides a variety of meters and tests that show you the instantaneous levels and spectral content of digital audio files with much greater accuracy and precision than is possible with hardware devices.**

will show a clip as short as a single sample whereas other systems (for example, the meters on a DAT deck) pass several clipped samples in a row before they acknowledge the problem. Sure enough, I found that the program nailed even a single clipped sample every time, and when set to its highest resolution, it didn't show any clips that weren't there.

The usefulness of the software's superaccurate clip indicator was borne out when I fed my system a sine wave that just clipped at the very top. *PAA*'s red lights blinked furiously while my TASCAM DA-30 mkII DAT was calmly telling me I had 3 dB of headroom left.

SPECTRAL DISPLAYS

To the right of the level meters is the spectral display, which divides the audio range from 43 Hz to 20 kHz into 24 bands, each approximately ⅓-octave wide. Individual vertical bars show the instantaneous level in each band. The display can show either the two halves of a stereo signal simultaneously or the summed mono signal. A switchable reference grid (called a Legend) shows amplitudes from -70 to -2 dB, and the display's gain can be raised by 6 dB or 12 dB.

There are peak-hold indicators here, too, with adjustable decay time. The main release time is adjustable, as is the display's update rate. Using a slow update and decay rate can give you a good picture of the overall spectral balance of a recording over time, which is very useful when you're mixing.

Below the spectrum is the waveform monitor, which will be familiar to anyone who has used a digital audio editor. It scrolls the waveform as the audio plays. The waveform monitor is more useful than most audio editors' displays in that it can show two channels in parallel, summed mono, or the difference signal (L - R). The displays can operate at any of three speeds or in reverse, and it's easy to zoom in or out on the time axis to see more or less detail. Any points at which the signal clips are displayed in red and stay that way. This feature gives you the clipping history of a piece.

PHASE SCOPE

The Phase Scope generates dots that represent the instantaneous stereo position of the signal against either a vertical or diagonal axis. You can choose

between two sizes of dots and three magnification scales. The Phosphor Emulation option causes the dots to die away slowly. A Length feature determines how long a dot stays on screen: at the longest length, the dots turn into little snakes. An overlaying grid can be switched on and off.

The update speed is not directly controllable, and because of an unfortunate limitation of TDM, it's not very fast. I found this feature to be the least useful of the bunch: it can give you an idea of the overall stereo spread of a mix and can show you when things are seriously out of whack, but if you're looking for the source of a balance or phase problem, the slow update speed makes it less than ideal. Apparently this display is given the lowest priority of the four. Although there's an improvement when all of the other displays are switched off, the Phase Scope is still no substitute for a simple oscilloscope set up to monitor left versus right.

GRIPES AND SUGGESTIONS

Although I highly recommend *PAA* for owners of fast Macintoshes who are serious about knowing what they are putting on tape or CD, there are a few ways the program could be improved. The owner's manual, written by Intelligent Devices President (and my fellow *Mix* columnist) Stephen St. Croix, has a tutorial that reads like one of his columns: breezy, clever, informative, and thought provoking, with plenty of amusing asides.

Unfortunately, there are no guidelines on what you need to run the software or instructions on how to set up your system. For seasoned Pro Tools users, this is not much of a problem, but it's likely to put off new Power Mac owners. There is neither a reference section, nor even a picture describing what everything does. The list of keyboard commands leaves out some of the most important ones, which you have to find by digging through the dense text. And the number of spelling mistakes is quite embarrassing.

Some less-visible features are clumsy and don't work too well. The command keys, used for setting many of the functions, can be reconfigured by the user, but you can't change just one command key: if you want to change any of them, you have to change *all* of them. Furthermore, the program does not show any difference between a

space and no assignment, so there's no way to tell if the space bar has been assigned to a function. The choices of command-key defaults also leave something to be desired. For example, the space bar turns the spectral display on and off, but in Pro Tools the same key is your main transport control, so when *PAA* is open, the key's transport function is disabled!

As with all TDM plug-ins, *PAA*'s window is always in the front in the TDM version. (The manual warns you that the plug-in won't work well if you put it in the background, but it doesn't mention that you *can't* put it in the background.) *PAA*'s large window makes this a particularly annoying problem. If you have a small monitor and want to do things such as operate Pro Tools' transport controls or bring *PAA* in and out of an insert point in the Mix window, you're going to have to slide things around a lot. You can't resize the window per se, but unlike other TDM plug-ins, *PAA* lets you choose to not display some sections, which frees up some screen real estate.

In a strange violation of Mac interface guidelines there's no title bar in

Product Summary

EM METERS	RATING PRODUCTS FROM 1 TO 5				
FEATURES	●	●	●	●	◖
EASE OF USE	●	●	●	●	
DOCUMENTATION	●	●	●		
VALUE	●	●	●	●	◖

the stand-alone version. (TDM plug-ins conventionally lack title bars.) To move the window in the stand-alone version, you have to hold down the Command key while you click and drag the window. Another anomaly is that you cannot shut down your Mac when the stand-alone version is running, as you can with other Mac applications; you have to quit the application first.

FINAL ANALYSIS

A simple description of its features really doesn't do justice to the great usefulness of this product. *Pro Audio Analyzer* can teach you a tremendous amount about how to mix music simply by allowing you to look at music in a wide variety of ways while you're hearing it. You may be tempted, as St. Croix says he was, to spend hours looking at other people's recordings to see why they sound good or not so good. You'll learn a lot by just playing CDs through it and watching the levels and spectral balances evolve. It would be time well spent. With any luck, you will learn to apply the principles you learn (the good ones) to your own recordings.

Rough edges aside—and I trust that some of these will be cleaned up in future revisions—*Pro Audio Analyzer* is a marvelous tool for anyone serious about audio. If you work with sound, you will find a use for it. Studio designers will use it to tune rooms, live-sound engineers will find hot spots and feedback points in performance spaces, mastering engineers will look for anomalies and imaging problems, broadcasters will make sure their signals are providing maximum punch, etc.

PAA makes it easy to maximize levels when mixing your own music, thereby optimizing your system's dynamic range without encountering the unconscionable horrors of digital clipping. It's faster and more critical than just about anything else out there, certainly anything that costs less than several thousand dollars. Paired with a good digital dynamics processor, such as Waves' *L1-Ultramaximizer*, you would have an unbeatable combination for making mixes that scream without screeching.

Paul D. Lehrman *is the coauthor of* MIDI for the Professional *(AMSCO) and writes the "Insider Audio" column for* Mix *magazine. He once wrote an audio analysis program called* Freud *for the Apple II.*

Upgrade to PowerTracks 3.5 for Windows $15

New features in Version 3.5... Enhanced Notation Window • Karaoke lyrics window • Notation scroll ahead • Notation clean mode • Music alignment to click track • Force all files to PPQ option • Improved hand splitting • Intelligent enharmonics

In the past, PowerTracks has offered powerful features such as music notation, wave file support, lyrics, chord symbols and score printout. Now we've added even more! PowerTracks 3.5 has a big "Karaoke Style" window to display lyrics up to "full-screen". The notation is greatly enhanced with support for finer resolutions such as 32nd notes. You can also now align a rubato piece to a click track. We've improved the intelligent piano hand splitting routines. The enharmonics are intelligently chosen based on the chord symbols, and more – Read on for more details.

Big Lyrics window lets you view song lyrics in the font of your choice in a full-screen window. Select your own font style, size, and color. Great for singing along with the music!

Notation Scrolling Ahead. If you're sight-reading the music off of the screen, you'll be pleased to note that the music can now scroll ahead. This is how we normally read sheet music! You can set the notation to scroll 1 or 2 bars ahead of the music *without* interfering with your view of the current notation. A great feature for sight-reading practice.

Rapid Resume. PTW can now automatically save the window positions of all opened windows and restores them the next time the program is used. You can optionally store and recall setups with a ".WS" file. Have PTW start out with your favorite setup every time you run the program by simply saving a file called FAVORITE.WS to your PT directory.

Save Your Patches. The enhanced patch select dialog now lets you select the patch list from within the dialog. PowerTracks 3.5 will remember the patch list used for each track and will save the name of the patch list used for each track to a ".SEQ" file. A "must have" for people who own multiple sound modules.

SysEx Input Buffer Size setting is now adjustable within the Options I MIDI. In dialog to help the people who normally have SysEx-input problems. The Buffer Amount can also be set.

Notation Enhancements:

- supports finer resolutions which allows you to enter notes such as 32nd and 64th notes
- "bars per line" markers which allows you to adjust the bars per line on a screen by screen basis
- "hard rests" which will be in effect even if you have minimize rests set to false, as well as the option to make notes invisible.
- a special mode for autodetection of finer resolution and/or multiple chords. PowerTracks can now "clean up" the kinds of notes & chords that can otherwise look like a bunched up mess.
- the Chord Note Separation and Chord Total Separation settings are now fully adjustable
- "Clean Mode" which automatically "cleans up" grace notes and any unwanted glitch notes from the notation. Glitch Velocity and Glitch Duration settings are fully adjustable.
- transpose setting will now transpose the key signature *and* the chords.
- improved Guitar Tablature draws the note and fingering exactly like your favorite magazine.
- improved Lyrics editing makes it possible to cut and paste using the Windows clipboard.

Align Music To Click Track command will align the music you've entered or recorded to a click track. This is very useful for songs that were recorded rubato or without using a metronome.

Improved hand splitting. Instead of arbitrarily splitting the keyboard into two parts, PowerTracks can make intelligent hand-splitting decisions which results in a highly accurate depiction of how the song should be actually played on the piano. These parts are also put on separate MIDI channels so that you can play or edit them individually.

Intelligent accidentals (enharmonics). sharps(#) and flats(b) are now context sensitive to the key *and* the chord its in. For example, if you are in the key of Eb and the chord is a D7, a F# note will show up as an F#, and not a Gb. This means that the notation accidentals will automatically show up correctly.

Convert loaded files to specific PPQ. Regardless of what PPQ a song was recorded in, you can work with it at the PPQ that is most familiar to you. For example, set this to 480 and all files will be set to 480, regardless of their original PPQ.

Forced Channels. channels that you normally use for different instruments (e.g. Bass on Channel 2) can be embedded in a midi file in addition to .SEQ files.

New Tips of the day. We have added some new tips of the day to keep you informed of the new features of version 3.5.

Hot Keys. Common procedures such as "Select all" and "Run DLL" can be performed with a single keystroke.

Other Enhancements such as a new setting to prevent PowerTracks from sending an "All Notes Off" command when synched to MIDI or SMPTE. This is important if you're using a lighting controller or other device that has a tendency to 'misbehave' when receiving an All Notes Off command.

Aardvark AardScape

By Dan Phillips

Analog warmth for a digital world.

Not long after getting my first open-reel, multitrack tape recorder, I made what might have been an egregious mistake: while recording the third and best take of a screaming guitar lead, I lost track of the level going to tape. When I looked at the meters halfway through the solo, they were jammed far into the red. As the guitarist was finishing the take, I started to mentally compose my apology. Before saying anything, though, I decided to listen to the overcooked solo.

The reader will already have guessed what I discovered: it sounded *brilliant*. Listening to the dense, rich, full tone coming back from tape, I tossed aside my half-formed apology and instead made a note to try that trick again. Of course, I had simply come across a common studio technique: using analog-tape saturation to add compression and a touch of pleasant distortion to guitar, bass, and drum tracks. This method uses the tape itself to create an effect.

Despite the many advantages of digital recording, this is one game it won't play. Hit a digital recorder too hard, and the sound changes instantly from precise, linear clarity to brittle, unpleasant distortion. Digital recording simply is incapable of generating the warm-sounding effects of analog-tape saturation.

But that doesn't mean you can't *add* those effects to a digital signal. Aardvark, a manufacturer of stand-alone A/D and D/A converters and digital audio synchronization equipment, has designed the AardScape ($595) just for that purpose. The half-rackspace, single-channel signal processor uses solid-state analog circuitry to simulate analog-tape saturation.

PANEL DECISION

The AardScape's front-panel layout is clean and simple, making basic operation quick and straightforward. The unit's three knobs—Input, Warmth, and Drive—each have ten detent positions, so it's easy to restore often-used settings. The knobs also rotate freely between detents for finer adjustments.

In addition to the knobs, there are two 3-position switches, labeled Saturation and Brilliance, that control the basic character of the tape-saturation effect. A bypass button, equipped with a bright red LED, allows you to compare processed and unprocessed signals. Input level is displayed in 3 dB increments on a 9-segment meter that includes a red "Ovr" LED. A power on/off button rounds out the front panel.

Although the controls are mostly easy to understand, the Drive knob initially confused both me and another engineer-producer. The name would seem to imply that the control affects saturation in some manner, but in fact, it's simply an output-gain control. Call me old-fashioned, but I'd like to see volume controls labeled as such. Leave the special names for special functions.

The AardScape provides both mono,

¼-inch unbalanced (-10 dBV) and XLR balanced (+4 dBu) inputs and outputs. A switch on the back panel selects between the balanced and unbalanced inputs. Both outputs are always active. Power is supplied via an external wall wart; I would have preferred a built-in transformer.

THE SATURATION PROCESS

The AardScape's saturation process utilizes a combination of compression and odd-harmonic distortion. As with traditional compressors and distortion boxes, the AardScape's processing is nonlinear; that is, the saturation effect changes in relation to the input level. The higher the input level, the more pronounced the effect.

The Saturation switch allows you to choose between Soft, Medium, and Hard nonlinear saturation curves, each of which delivers a different mixture of compression and overdrive. The Warmth knob functions as a fine-tune control for the Saturation switch. When Warmth is turned all the way up, the saturation is at maximum intensity for the current setting; with Warmth turned down, the effect is almost bypassed.

The Brilliance switch offers three EQ variations: Full, Clean, and Brite. The Full setting evidently bypasses the EQ, but when combined with moderate to high amounts of saturation, it produces a fairly bass-heavy sound. The Clean setting tightens up the bass response, but the resulting tone is still warmer and darker than the original signal. I almost always preferred the Brite setting, which combines the Clean setting's tighter bass with a restored high end.

Because the EQ is in the signal path prior to the saturation section, its settings alter the compression/distortion characteristics as well as the overall tone. The Full setting, in particular, seems to drive the saturation much harder than the other two choices.

SATURATED PHAT

I used the AardScape in my ADAT-based studio over several months on projects ranging from acoustic ballads to techno/industrial dance tracks. Some friends also used it during mixdown on a guitar-based alternative rock album.

The device worked well on male, alternative-rock vocals. On the techno/industrial track, I processed the

Aardvark's AardScape uses compression and odd-harmonic distortion to simulate the sound of analog-tape warmth.

● **A A R D S C A P E**

lead vocals through the AardScape's Hard and Brite settings to create a heavily squashed, distorted, crunchy, and angry tone, which worked perfectly. On a ballad vocal, however, I had less luck: even the Soft setting produced a bit too much distortion for my taste.

For the alternative album mix, my friends used the Soft and Clean settings on the lead vocals to create what they termed a "P.A. simulator," which added a fair amount of compression combined with a slight amount of distortion. They felt that the AardScape made the vocals sit better with the guitars, making it possible to reduce the volume of the vocals without losing them in the mix.

On heavy, electric "crunch" guitar, the AardScape came close to duplicating the effect of my happy accident with the analog multitrack. The Medium and Hard settings produced overdrive and compression timbres that were quite complementary to the existing guitar tone. When used on a cleaner, jangly rhythm guitar, though, the AardScape overaccentuated pick noise.

RHYTHM TRACKS

I particularly liked the way the AardScape dealt with kick drums, giving them a full, round sound. This was the instrument on which the processing sounded the most like real analog-tape compression. Snare-drum signals also

fared well, emerging from the Aard-Scape with a fatter timbre and a nicely emphasized buzz of the snares. However, the effect also included a certain amount of transient smearing. This sometimes resulted in certain instruments—notably hi-hats—not sounding as crisp as I would have liked.

On a mono submix of an entire drum kit, I had fun driving the AardScape's inputs very hard to produce a truly messed up, pumping and throbbing beat, morphing what had been a clean drum sound into something approaching the timbre of a sampled, vintage-drum loop.

I also tried processing a number of sampled bass sounds but got mixed results. For example, a fuzz bass sounded absolutely spectacular after a trip through the AardScape, with both increased presence and a beefier low end. On the other hand, a clean, acoustic-bass sample became too gritty. Likewise, simple analog-synth basses seemed to benefit, but whenever I tried a sound with filter resonance, the distortion component of the saturation process became too strong.

SATURATED SOLUTION

The AardScape is a unique-sounding processor. On kick drum, crunch guitar, and male rock vocals, it delivers a pretty good simulation of analog tape saturation. On other instruments, the effect may not be a dead ringer for tape saturation, but it's often interesting and useful nonetheless. It tends toward the heavy side, though. You'd have to hit analog tape pretty hard to approach the AardScape's level of saturation.

As the AardScape is a mono unit, the list price of $595 seems a bit steep. But if you like the unit so much that you need two of them, Aardvark offers a calibrated pair, complete with a rack-mount tray, for $1,095. Although I didn't get a chance to test a calibrated pair, it would be the way to go for processing a stereo mix.

It may not be a panacea for all possible source material, but the AardScape is useful for several applications. If you're looking for a way to bring some analog life to your digital recordings, give this box a listen.

Dan Phillips, a singer-songwriter in the San Francisco Bay Area, is a principal in Touch Productions, a firm providing music for television, film, and multimedia.

Product Summary
PRODUCT:
AardScape tape-saturation simulator
PRICE:
$595 ea.
$1,095 for dual-mono pair
MANUFACTURER:
Aardvark
tel. (313) 665-8899
fax (313) 665-0694
e-mail lovell@
aardvark-pro.com
Web http://www
.aardvark-pro.com
CIRCLE #442 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5				
FEATURES	●	●	●	◖	
EASE OF USE	●	●	●	●	
AUDIO QUALITY	●	●	●	●	
VALUE	●	●	●		

WRH

By Gerry Bassermann

Spatialize your
sampler with
3-dimensional sounds.

3D Sound's *3D Keys, Vol. 1: The Dream Weaver* ($199) is an eclectic sample library that features Gary Wright's Fender Rhodes and synth sounds along with solo viola, assorted digital-synth confections, explosions (for mixing with kick or snare drum), and even a snarling dog. As if that weren't enough to distinguish this CD-ROM, all sounds are sampled using 3D Sound's proprietary 3-dimensional sampling technique. (A warning on the tray card reads, "Caution: Some sounds on this disc may cause dizziness.")

Dream Weaver is available for the Kurzweil K2000, Akai S1000/S3000, E-mu EIII, and Digidesign SampleCell II.

I checked out the collection on an E-mu EIIIxp with 32 MB of memory. The disc contains eleven sound banks for a total of 344 MB of data. Some banks consumed the entire memory, but it is possible to load single presets, and only a few patches required the full 32 MB. The E-mu EIIIxp format is the most basic and limited version of the disc; in contrast, the K2000 has more banks, with twice the number of presets, and offers more control of touch, filters, and envelopes. Unfortunately, I did not get a chance to review this format.

PROCESSING SPACE

3D Sound specializes in creating soundscapes that sound 3-dimensional when played back on two speakers. The process, according to 3D Sound Director Franz Pusch, uses an enhanced form of binaural recording. Typically, binaural recordings (made with an artificial human head fitted with mics in place of ears) sound great on headphones but exhibit phase anomalies when played through stereo speakers. 3D Sound's process purportedly compensates for these phase problems.

The result is a resonant, full, and spacious sound that comes closer to simulating real space than reverb algorithms do. Naturally, the extent of the effect depends greatly on the quality of the acoustic environment and the listener's position in the stereo field. But on headphones, the effect is amazing. (I don't know how many times I rose to answer a telephone that wasn't ringing!) Thanks to the spacious, 3-dimensional sound of *Dream Weaver* patches, I rarely felt the need to process them further with outboard effects.

DREAM ANALYSIS

To avoid the initial seduction of the binaural experience, I listened to the sounds through the speakers first and then through the headphones. This approach yielded unexpected results: as I listened through the speakers, I was impressed by the large soundfield and the "fresh" quality of the sounds. Listening on headphones, though, I discovered sonic blemishes and some less-than-perfect sample editing. For example, what sounded over the speakers like a velvety sheen on a synth patch revealed itself on the headphones to be a lot of noise in the sound files. According to 3D Sound, the noise originates in the keyboards that were sampled and makes the sounds more authentic.

Dream Weaver opens with three banks of stereo-phased, 73-key Fender Rhodes electric piano, one with fast phasing and two with slow. The pianos are sampled using the onboard stereo phaser played through an original Fender stereo speaker cabinet. The result is a resonant, glistening sound.

Each Rhodes bank contains two or three presets. The size of the preset is indicated by numbers at the end of the title. For example, "RhsSlowVib Hd06" indicates a 6 MB version of the Rhodes electric piano, played hard and with slow vibrato.

The presets offer extreme amounts of touch control. Strangely, though, the samples are not looped, so voices sometimes cut out while you're holding down long chords. There are also some rather startling timbral and speed-shift changes from one sample zone to the next. (The samples are placed roughly a tritone apart.) Fortunately, this isn't as noticeable during pedaled, polyphonic playing.

A fourth Rhodes bank is dedicated to octave samples. These sound really

3D Sound's *3D Keys, Vol. 1: The Dream Weaver* features 3-dimensional samples of Gary Wright's Fender Rhodes piano and assorted synths as well as such anomalies as cello, banging heads, and snarling dogs.

big and don't compromise polyphony because the octaves don't require two layered samples.

Another bank, "Leslie Organ," provides gorgeous, resonant, dynamic presets with slow and fast Leslie motion in the samples themselves. Just when you think these sounds are about as fat as can be, along comes a chorused version that sounds even fatter.

The "3DKeys Percuz" bank features distinctive-sounding marimba and vibes. Even if you have plenty of keyboard-percussion sounds in your collection, you probably don't have any that sound like these. The only problem is that the samples are quite noisy. Fortunately, the noise is masked by tight envelopes.

"SuperWide3DPads" are just that, but again, the amount of noise in the samples, along with a high resonance factor, compromises their potential. The star of the bank is "3DreamPadTheN," a synth string pad with high digital overtones.

The "Dream Filters" bank has the most interesting and 3-D sounding files on the disc. The voicing is particularly

inventive, with cascades of synth and FM-like sounds. There's even a good dose of analog kookiness *à la* Louis and Bebe Barron, sound designers for the movie *Forbidden Planet*. If you're a sound designer creating commercial spots, one of these effects just might send your client to the moon.

"Mean 3D," which combines snarling dogs and head bangs, would make a perfect load for a horror-movie graveyard scene. The source material sounds truly scary (check out "EkoFeedX"). There's also a cool-sounding effects percussion patch. I only wish that each sample had its own preset; on the EIIIx, such transpositions can be wonderful fodder for industrial sound design.

The two "Dream Pad" banks that complete *Dream Weaver* offer 3-D versions of LA-synthesis clichés, namely string, vocal, or synth pads with icy harmonics wafting above. Unfortunately, there is absolutely zero preset programming—no Velocity sensitivity and no long, washing envelopes—even though the textures cry out for it.

FINAL INTERPRETATION

There are many sonic jewels in this collection and even some hidden treasure. I discovered one of my favorite sounds while scrolling through the file directory of Bank 07, "3D Keys [hordal" *(sic)*. Tucked away in a corner where you might easily miss it (along with evanescent bell trees, swirling bowls, and thunder strums), "02TBaUpSlide" features an incredible rising strum effect. Unfortunately, this marvel was so hard to find that I had to go back and re-

view each of the presets in order to relocate it. This is another example of a sound that should have been completely transposed across the keyboard. That way, it would be readily available to the studio player who doesn't always have the time to investigate file directories.

Because of its "read only" nature, the CD-ROM format demands a high level of quality control. Unfortunately, the programming on this CD-ROM is uneven. Though some of the mistakes

▼

I rose to answer a telephone that wasn't ringing.

burned into *Dream Weaver* are relatively insignificant, they are unfathomable nonetheless. For example, there are files that consume memory but aren't mapped to any preset. Also, some files are sampled at 44.1 kHz whereas others are sampled at 48 kHz. (According to 3D Sound, some of the source sounds did not translate well at 44.1 kHz.)

Other errors are more consequential. The Rhodes "RhsSlowVib Hd16," for example, is a beautiful and expressive preset, but keyboard zone F1 to D2 on that preset doesn't respond to Velocity, which makes the sound inconsistent. And the "Pleiades" preset in the "Dream Filters" bank, though otherwise strong, is mapped on the keyboard a whole step sharp. (This was intentional; the sounds slide down a whole step when sustained.)

As an archive of resonant, 3-D sound files, *The Dream Weaver* offers a diverse selection of often excellent and fresh sonic material. It is especially well suited to sampler programmers who enjoy creating custom sound designs. On the other hand, *The Dream Weaver* offers little in the way of expressive presets that are ready to load and play with no tweaking. Players will simply have to edit and save their own presets. However, their efforts will be well rewarded, as they will command an arsenal of fat, distinctive, 3-dimensional sounds.

Gerry Bassermann is a working musician in the San Francisco Bay Area.

Product Summary
PRODUCT:
3D Keys, Vol. 1: The Dream Weaver CD-ROM
PRICE:
$199
MANUFACTURER:
3D Sound
tel. (714) 643-7666
fax (714) 643-7615
e-mail fpusch@aol.com
Web http://www .3dsound.com/3d
CIRCLE #443 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5				
AUDIO QUALITY	●	●	●	●	
DOCUMENTATION	●	●	●	●	●
VALUE	●	●	●		

Lexicon PCM 90

By Richard Chycki

Lexicon still does digital reverb right.

These days, the Swiss Army knife approach has become the *de facto* standard in effects processors. Indeed, it is commonplace to find a dozen or more simultaneous effects in a single algorithm. But at least one processor bucks this trend: Lexicon's PCM 90 does reverbs and nothing but reverbs, and it does them extremely well.

Borrowing reverb algorithms from Lexicon's family of processors, including the 480L, 300, PCM 60, and PCM 70, the PCM 90 focuses solely on top-notch environment simulations. The unit melds 250 presets and five basic algorithms with an efficient user interface and extensive dynamic and MIDI modulation capabilities.

TWINS

If you were to glance at a PCM 80 (reviewed in the May 1995 **EM**) and PCM 90 in a rack, you would be hard-pressed to tell them apart. However, the PCM 90 has a few additional rear-panel features. Balanced XLR analog audio outputs are a welcome addition to the balanced ¼-inch TRS outputs. The analog audio inputs are Neutrik Combo jacks, which accept both XLR and ¼-inch TRS connectors.

An input-level switch adds 20 dB of gain; when engaged, it also switches the inputs from balanced to unbalanced operation. The output level is software selectable between +4 dBu and -10 dBu.

Digital audio in S/PDIF format can be sent and received by the unit (at a sample rate of 44.1 or 48 kHz) via RCA jacks. Unlike the PCM 80, the PCM 90 does not let you mix digital and analog inputs; you must select one or the other. However, the output appears at both analog and digital connectors simultaneously.

In most cases, the PCM 90 can process AES/EBU signals via the S/PDIF ports (with appropriate cable adapters). I successfully inserted the PCM 90 between an Apogee AD-1000 A/D converter and two DAT decks: a TASCAM DA-30 and a Panasonic SV-3700. However, Lexicon has acknowledged that some older equipment might have problems with mute and emphasis flags in the AES/EBU digital signal from the PCM 90. As a result, the manufacturer officially lists the unit as compatible only with S/PDIF.

An additional pair of ¼-inch TRS jacks lets you control various parameters with your feet. One jack can accommodate two momentary footswitches, and the other jack is intended for a continuous foot controller, such as a volume pedal. These inputs can be assigned to internal parameters or MIDI Control Change messages, which are transmitted via the MIDI output. This MIDI information can be used to control other MIDI devices, or it can be recorded to a sequencer for automated playback.

The PCM 90 uses the same 2-line × 20-character, highly visible alphanumeric display found in the PCM 80. A PC (PCMCIA) card slot accommodates ROM and RAM cards for preset storage and algorithm upgrades. Like the PCM 80, the PCM 90 uses the Select and Adjust "soft knobs" that first appeared on the PCM 70. Dual 5-segment LED meters monitor headroom for both channels. Input level is adjusted with a single control. Input balances and routing are programmable where applicable (e.g., the Chamber/Room split algorithm).

The PCM 90 uses a pair of Lexicon's flagship Lexichip IIs to perform its reverb computations whereas the PCM 80 uses a combination of a Motorola 56002 DSP and a Lexichip II. The PCM 90 contains an international power supply that can be connected to any AC line from 100 to 240 volts with no switching required. The outputs are muted during power-up and power-down.

PRO AND GO

An efficient row-and-column matrix organizes all presets, user programs (called registers), and edit parameters (see **Fig. 1**). The Up and Down buttons select rows, and the Select knob specifies the column. Occasionally, a master parameter, such as a master delay, has a submatrix that is accessed by pressing the Load button. Whenever possible, parameters remain in the same matrix location from algorithm to algorithm.

The PCM 90, like the PCM 80, has a multilevel user interface. Pro mode provides access to all parameters within the matrix. This mode is best suited for writing user programs and meticulous editing. Go mode provides access to a single "soft row" of ten parameters, which are selected by the user from the full edit matrix in Pro mode. Go mode makes for efficient use in the field, where speed is often a significant issue but a moderate level of editing may be necessary.

For even more efficient operation, one primary parameter can be controlled with the Adjust knob. Don't be fooled by the concept of a single parameter. It's actually a powerful aggregate of up to ten parameters from the master matrix coupled together. You can assign any parameters from the master matrix to be controlled by the Adjust knob.

Any delay-based parameter or time-based modulator (for example, LFOs or switches) can be assigned a tempo value by simply pressing the Tap button in time with the music. Tempo is

The PCM 90 offers that famous Lexicon reverb sound and powerful dynamic modulation capabilities in an easy-to-use package.

● **P C M   9 0**

FIG. 1: Each algorithm offers a matrix of editable parameters. (Courtesy Lexicon, Inc.)

automatically averaged over two to eight taps (user-selectable), and it can be set manually for a preset or globally. The tempo range is 40 to 400 bpm. The PCM 90 can derive tempo from incoming MIDI Clock messages, and it can function as a MIDI Clock master, generating and transmitting its own MIDI Clock to other MIDI devices for synchronization purposes.

The PCM 90 offers a ton of LFOs, switches, and MIDI modulation capabilities. Almost any parameter can be modulated, but you must remember that modulating some parameters causes audible artifacts. In some cases, the unit mutes its output during parameter changes. For example, changing Reverb Size mutes the unit whereas changing High Cut causes no interruption.

Up to ten modulators can be en-gaged per preset, using a scheme for selecting source, destination, and modulation range that is similar to the one found in the PCM 70. Judicious modulation adds depth and life to reverbs. In addition, the PCM 90 includes a number of parameters, such as Wander and Spin, that add natural, random movements to the reverb tail.

A little delay modulation goes a long way toward softening the surgical precision of the digital world. The envelope follower is a particularly powerful modulator. When assigned to reverb time or brightness, it can add accents that track the dynamics of the input signal.

An edit buffer allows you to quickly switch between the current edited program and the factory preset from which the program is derived. The bypass
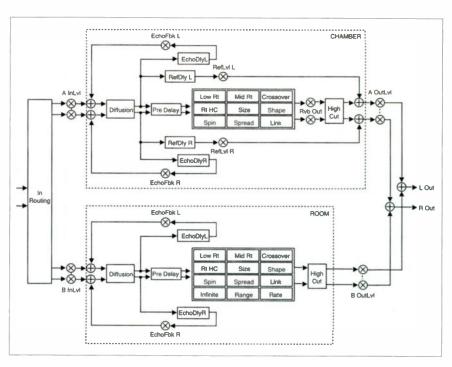
FIG. 2: The Chamber/Room algorithm is the only dual effect in the PCM 90. (Courtesy Lexicon, Inc.)

function can be configured to mute the input or output or pass the dry signal. Muting the input allows the reverb tail to decay smoothly and saves the labor of assigning the Input Level parameter to a switch.

REVERBERATOR

The PCM 90 uses five internal algorithms for its reverb effects: Plate, Ambience, Random Hall, Concert Hall, and a split Chamber/Room. All are true stereo effects, not just discrete dual-channel processing. A generous number of delays in each algorithm make additional echo and chorus effects possible, but the unit's parameter architecture is optimized for reverbs. Expect to use some programming chops if you care to walk that path.

Small spaces are notoriously difficult to simulate electronically: ringing and harshness are common negative artifacts. The Ambience algorithm in the PCM 90 is a tribute to Lexicon's experience in the digital-reverb market. This algorithm imparts a natural reflective quality that is surprisingly easy to mold into a variety of truly realistic environments.

For example, I often record a guitarist who insists that his solos be recorded with plenty of room ambience. The room in which we work is small but quite live. Digital reverbs are a no-no; he needs *real* room. We generally use three tracks: one dry and two for ambience. With the PCM 90 and a few minutes of tweaking, I was able to simulate his guitar ambience so well that it passed scrupulous solo and headphone listening tests.

Product Summary

PRODUCT:
PCM 90 digital reverb
PRICE:
$2,995
MANUFACTURER:
Lexicon, Inc.
tel. (617) 736-0300
fax (617) 891-0340
e-mail 71333.434@
compuserve.com
CIRCLE #444 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5				
FEATURES	●	●	●	●	
EASE OF USE	●	●	●	●	
AUDIO QUALITY	●	●	●	●	◖
VALUE	●	●	●	●	

PCM 90 Specifications

Analog Inputs (XLR/TRS)	2/2
Analog Outputs (XLR/TRS)	2/2
Digital Inputs/Outputs	1/1
Digital Audio Format	S/PDIF
Factory Presets/User Registers	250/100
A-D Converters	16-bit, 64x oversampling
D-A Converters	18-bit, 8x oversampling
Sample Rates (kHz)	44.1, 48
Frequency Response	10 Hz–20 kHz (±0.5 dB)
THD (A/A*)	≤0.008%
S/N Ratio (A/A*)	≥90 dB (20 kHz bandwidth)
Dynamic Range (A/A*)	93 dB (20 kHz bandwidth)
Operating Levels	-10 dBu, +4 dBu

*analog input to analog output

I didn't restrict this algorithm to guitars, either. By turning down the diffusion and using a nominal room size, I achieved obnoxiously wide, hard-sounding drums that seemed unprocessed. This effect worked great for spreading out solo and group vocals, too. Using the "HardwoodRoom" preset, I was very successful in matching some vocal punch-ins done in a dead environment with a basic track performed in a live, wooden room.

The Concert Hall algorithm is a revamped version of a similar algorithm in the venerable PCM 70, with some significant perks: a digital compressor/expander and spatial EQ. Reminiscent of the digital compressor in Lexicon's 20/20 A/D converter, this compressor boasts much more severe limiting capabilities. Available compression ratios of up to 100:1 can be applied to the reverb in order to mutate the reverb tail.

High compression levels with correspondingly high gain settings can result in clipped transients, especially on percussive program material. The PCM 90 can circumvent this situation by means of a lookahead function on the compressor. In effect, the compressor can activate before the transient arrives. Of course, this delays the signal, but the size of the lookahead buffer (and thus, the length of the delay) is user-definable. Under most conditions, the delay should not be perceptible.

The spatial EQ manipulates the apparent spaciousness of the stereo signal in an unusual way. A process called "negative cross-feed" subtracts up to 6 dB of the left channel's low-frequency information from the right channel and vice versa. (The low-frequency threshold is user-definable.) Positive cross-feed adds left-channel low frequencies to the right channel and vice versa, which helps bring bass parts toward the center.

The inherently slow density build of the Ambience algorithm, coupled with copious compression and low-end enhancement, can create truly fierce, barking reverbs that walk the line between reverse, gated, and big-size/short-time reverbs. These settings add tremendous weight and presence to snares, kicks, or toms. The EQ and dynamics features quickly become addictive and would be a welcome addition to the other four algorithms.

The Plate algorithm offers a characteristically dense, diffuse reverb with a number of echo possibilities in series and parallel with the reverb. Delay density and diffusion are very high in this algorithm, creating a plate simulation that is useful on many recorded sources, from drums to vocals to acoustic guitars. Presets such as "Floyd Wash" and "Vocl EkoPlate" capitalize on the liquid echo/reverb combinations possible in this algorithm.

Random Hall is the meat-and-potatoes algorithm of the PCM 90. Emulating both halls and rooms, this algorithm reminds me of the rich Chamber algorithm in the PCM 70, albeit substantially more refined and with greater editing control.

The Chamber/Room algorithm is a solid indicator of the PCM 90's processing potential (see **Fig. 2**). This dual algorithm allows full stereo processing

(or mono in/stereo out) with the output of the room and the chamber summed to the stereo outputs. Neither the room nor the chamber displays any sonic shortcomings from the dual processing. Versatility within the algorithm is not compromised either.

The large number of parameters makes this algorithm moderately page intensive in Pro mode. In addition, both the chamber and room must share the ten available modifiers. These are minor consequences, though, given the level of processing offered.

In mixing applications, I often set up this algorithm in split mode: input A feeds the chamber and input B feeds the room (configured as mono in/ stereo out). The chamber/room combination worked well for kick and snare, especially for contemporary, small drum rooms. I heard nary a trace of ringing. Presets such as "Two Chambers" and "LgKick/Snare" are indicative of this dual application. Radically different drum effects, such as individual equalization and compression, are difficult to achieve due to the summed output.

The Chamber/Room algorithm is the only combo effect available in the unit. It made me want to try different combinations of other reverbs, such as plate and room, two plates, or two rooms. At the last Winter NAMM show, Lexicon announced that they are developing an algorithm card for the PCM 90, which is scheduled to be released by the end of 1996. This card will contain ten new algorithms, including five splits, four cascade or serial effects, and one surround-sound algorithm.

The manual is particularly useful for designing your own reverb programs. It clearly defines all the parameters within each algorithm and indicates guidelines for using them in addition to the pros and cons of various parameter settings. The laminated quick-setup card is a nice touch.

PRESETS
The PCM 90 comes with 250 onboard presets, organized into five banks of 50. The banks organize the presets into halls, rooms, plates, post-production presets, and split presets. Each bank is also subdivided into applications, such as vocal, instrument, and live sound. Up to 100 user programs can be stored onboard in two banks of 50.

To help you select a preset, up to four keywords can be assigned to each one from a list. By selecting a keyword from the list and pressing the Program or Register button until the Keyword page appears, you can rapidly scroll through all presets and user registers containing that particular keyword. This is particularly handy when the on-board and card memories are full; more than 1,000 programs can be stored on a 1 MB PC card.

With most equipment, my tendency is to select a preset and really dig in to customize it for my situation. With the PCM 90, the majority of presets are eminently usable, and the keyword system goes a long way toward helping you select a proper preset that closely matches the application. Often I simply dialed up a preset and tweaked the Adjust knob to slot it into my mix.

This is a testament to the operating system and the attention paid to the presets. Many of the presets have unique metaparameters assigned to the Adjust knob. For example, "PCM 60 Room" has four reverb times assigned to the Adjust knob, appropriately labeled by LED color as used on the PCM 60 push-button matrix.

CONCLUSIONS
Lexicon's background in digital reverb is second to none. This is apparent in all the company's products, which combine an appealing user interface with world-class reverbs that boast the famous Lexicon sound. The PCM 90 is no different.

Appealing to the widest possible spectrum of users, the Pro and Go approach is an effective compromise that should satisfy most users, from the most experienced tweakers to those who hunger for instant gratification. The programming and modulation potential is formidable in this unit; you're sure to come up with truly personalized effects.

In these days of rapid obsolescence, Lexicon's intention to release more algorithms that expand the PCM 90 capabilities is commendable and displays the company's support of the equipment's open architecture. And if you combine the PCM 90 with the PCM 80, you get a hard-to-beat sonic palate of effects.

Richard Chycki is a producer-engineer-guitarist in Toronto, Canada.

Z-Systems z-8.8

. . . .

By Larry the O

**Finally,
an affordable digital
patch bay.**

Digital inputs and outputs used to be exotic, expensive, and esoteric features. Accustomed as we now are to once-inconceivable advances becoming commonplace, it should be no surprise that most project studios and full-service professional studios have several, and often many, pieces of equipment with digital I/O.

Three such devices can be connected in any configuration simply by moving cables around. But once you have more than three units with digital I/O, moving cables quickly becomes too awkward to be workable. With a DAT deck, sampler, and digital audio workstation, you're nearly there. Throw in an effects processor or CD deck, and you're over the top. At that point, the need for a digital audio patch bay or router is obvious. This equipment has long been an expensive proposition, more than most home and project studios could afford or justify. But not anymore.

Z-Systems makes expensive, high-end digital audio routers and some nifty utility boxes, but with the release of the z-8.8 Digital Detangler, the company managed to push the price of an 8-input by 8-output digital audio router down to the $1,000 range. Suddenly, another key to sanity is within the home-studio owner's reach.

A CONNECTION CORRECTION

The z-8.8 mounts in two rackspaces and can be ordered in either of two standard configurations: all AES/EBU or four AES/EBU, two coaxial S/PDIF, and two optical S/PDIF. Custom configurations are available with any combination of AES/EBU or S/PDIF (coax or optical) input/output pairs. The review unit was configured (at my request) with four AES/EBU and four coaxial S/PDIF ports.

All connections are on the rear panel. AES/EBU connections appear on XLR connectors whereas coaxial S/PDIF inputs and outputs are on BNC

connectors. This is bad news: although Sony's original SDIF digital I/O scheme employed BNC connectors, the IEEE 958/CP 340 standard that defines the interface we know as S/PDIF clearly specifies RCA connectors. For the review, Z-Systems sent me eight Radio Shack RCA-to-BNC adapters. I felt a little silly connecting my gold-tip, low-capacitance RCA cables to cheesy Radio Shack adapters that, as it turned out, gave me intermittent problems. Also on the back panel is a standard IEC detachable AC power socket.

The good news, however, is that, at the time of writing, Z-Systems had located RCA jacks compatible with their mechanical design, and the z-8.8 should be shipping with RCA connectors for S/PDIF by the time you read this. (That wasn't the only improvement made to the z-8.8 during the review process, but we'll get to that in a moment.)

When an S/PDIF input is routed to an AES/EBU output (or vice versa), the conversion is simply an electrical conversion; no flags are changed. Some early S/PDIF implementations do not successfully convert to AES/EBU in this way, but all of the more recent gear I tried worked with no problem.

NO MUSS, NO FUSS

The front panel centers around a simple set of controls: four buttons (Cancel, Load, Save, and Route), two knobs, and an LED display. The rather small LED display shows the current connection scheme. The legend beneath the display indicates the numbers of the outputs; the display itself shows which input is connected to the output indicated below. The labeling is simple and unambiguous: the display is labeled "From (Source)," and the legend below says "To (Destination)." That's easy. Similarly, the two detented knobs are marked "To" and "From." That's pretty straightforward, too.

To connect an input to an output

you need only turn the To knob until the display above the desired output legend is flashing and then turn the From knob until the desired source is shown in the flashing display. Now your connection is set up. Press the Route button, and your connection is made. It's a real no brainer, which is exactly what a router should be. As an electronic router, the z-8.8 allows an input to feed any number of outputs simultaneously, but each output can only be fed by a single input.

Loading preset configurations is easy: press the "load" button, use the To knob to dial up the configuration you want (which is shown on the display as you scroll through), press Load again to confirm your choice, and hit the Route button to make it active. Saving is even easier: press Save, dial in the preset where you want to put it, and then press Save again to store. In either case, you can press Cancel if you decide to back out halfway through. If you press the Route button, the patch shown is written to preset 00. Whatever is in this preset is what gets loaded on power-up.

And that, my friends, is a complete functional description of the z-8.8. I find it hard to believe anything with a computer inside it could be this easy.

PRACTICAL APPLICATIONS

I used the z-8.8 while doing sound design for LucasArts Entertainment computer games. My rig includes a CD player, digital audio workstation (DAW), sampler, DAT recorder, and signal processor, all of which have digital I/O. I use about a half-dozen patching configurations on a regular basis: CD player to DAW, CD player to sampler, sampler to DAW and vice versa, etc.

With my first review unit, I was pleased I only had to punch buttons to repatch instead of moving cables. (The first unit I tested used a button for assigning each output instead of the two knobs.) But I soon wanted to

Z-Systems' z-8.8 Digital Detangler is easy to use, works great, and is the most affordable digital audio router we've seen.

store presets of common patches, which the first z-8.8 couldn't do. The new units can save up to 80 configuration presets, which is a big improvement.

When I started my tests, my DAW was an old-style, 4-channel Digidesign Pro Tools system that allowed me to use its digital output as a stereo effects send and simultaneously record from the digital input. With a couple of knob turns on the z-8.8, I could put my Lexicon PCM 80 effects processor into a digital effects loop! It was way cool to process sound effects and have them end up as files on my hard disk. (My Pro Tools Project system does not allow this. I'm bummed.)

I found it useful to make up some paper labels indicating which piece of equipment was connected to each input/output pair and what each of the presets did. I simply taped these to an open section of the front panel for easy reference. Up to this point, I've only needed about seven presets because some presets can serve double duty. For example, preset 01 on my unit has the CD player patched to both my DAW and my sampler. I can record from a sound-effects CD into either recording device using the same preset.

PATCH ME THROUGH

The Z-System's z-8.8 Digital Detangler is amazingly simple, works just dandy, couldn't be any easier to use, and costs way less than anything else that does this job.

The manual is only three pages long (four, if you count the warranty), but it covers everything. Who'd believe it?

I suppose I could start spouting a list of features I'd like to have, but they'd all push up the price. If pressed, I'll admit that the display could be a little larger. But if you really need advanced functions, you're doing serious enough work to warrant a more expensive router. With two rows of XLR connectors on the back, I can't even expect them to find a way to put the unit in a single rack space.

The bottom line is simple. If you need fast, effective, basic routing of digital audio on a project-studio budget, buy the z-8.8 Digital Detangler.

In addition to producing sound design for computer games at LucasArts Entertainment, **Larry the O** *actively performs and records with Annwn and 11:11.*

Product Summary

PRODUCT:
z-8.8 Digital Detangler digital audio router
PRICE:
$980 w/all AES/EBU
$1,080 w/4 AES/EBU,
2 coax S/PDIF, 2 optical S/PDIF
$1,325 for custom configurations
MANUFACTURER:
Z-Systems Audio Engineering
tel. (352) 371-0990
fax (352) 371-0093
e-mail: z-sys@z-sys.com
Web http://www.z-sys.com
CIRCLE #445 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5				
FEATURES	●	●	●	●	◖
EASE OF USE	●	●	●	●	●
DOCUMENTATION	●	●	●	●	●
VALUE	●	●	●	●	◖

Folded Space Micro Room

By Richard Chycki

Turn your guitar up to eleven without disturbing the neighbors.

Guitarists, in their quest for the ultimate tone, often resort to overdriving their vacuum tube-based amplifiers. Unfortunately, the effect of this action is usually massive volume. That's very bad news for the neighbors if your personal studio resides in an apartment—especially if the creative bug strikes at 3 A.M.

Obviously, most small home or apartment studios cannot play host to 3-figure decibel levels. The Micro Room by Folded Space Technologies is designed as an answer to this common predicament. Quite simply, the Micro Room is a speaker and microphone combi-

Folded Space Technologies' Micro Room is an enclosed microphone/speaker system that allows you to record raging amps without raging volume. Although it doesn't sound like a 4 x 12 cabinet as the manufacturer intended, it has a useful, distinctive sonic signature.

nation sealed into an enclosure that lets you plug in your favorite amp (up to 100 watts) and unleash a sonic barrage without disturbing your neighbors. For those guitarists who are not impressed by digital speaker emulations, the Micro Room offers an interesting alternative for tracking raging guitars.

INDIRECTLY SPEAKING

I received two test units over the course of this review. The first unit was jarred in transit, and after some testing, I determined that it was not quite suitable for review due to the possibility of slight microphone misalignment. Rather than sending an identical replacement, the manufacturer took a number of suggestions that were made for the first unit and sent us a "new and improved" model for review.

The standard Micro Room contains a 5.25-inch, 8-watt, Folded Space MR-525 speaker miked with a Shure SM57. The manufacturer states that the all-paper driver was selected to sound like a vintage 12-inch speaker and that the sealed speaker/enclosure combination is tuned to emulate the response of a 4 × 12 cabinet. (A ported version is available that emulates an open back, 1 × 12 enclosure.) Any driver up to eight inches in diameter may be substituted for the factory speaker, although the enclosure has been optimized for drivers smaller than six inches in diameter. All options and modifications are priced and completed on a per-job basis. The manufacturer offers a $75 credit for the SM57 and a $25 credit for the speaker toward a microphone and driver of the purchaser's choice.

The hardwood enclosure measures 12 × 12 × 22 inches and is protected with a textured, black epoxy finish. However, to survive heavy use on the road or in the studio, the Micro Room would have been

well served by metal caps on the cabinet's corners and rubber feet on its bottom surface. A standard metal roadcase handle is mounted on the side of the unit; the XLR connector for the microphone and ¼-inch high- and low-power speaker jacks are set into the handle plate.

The stock Micro Room's high-power input is designed for use with amps that deliver 30 watts or more; the low-power input is for use with less powerful amplifiers. Upon the purchaser's request, Folded Space will set your Micro Room for use with more- or less-powerful amps.

The mic is accessed by removing a wooden end panel. The mic compartment is filled with layers of 2-inch acoustical foam; the speaker section is left untreated. The SM57 is firmly clamp-mounted over the center of the driver.

Folded Space encourages experimentation with mic placement. Changing the mic's distance from the driver is simply a matter of loosening the thumb screws on the clamp. Laterally moving the microphone requires removing and repositioning the clamp mount. I found that changing the distance of the microphone from the driver produces a more dramatic sonic effect than changing the mic's lateral position, so the hassle of repositioning the clamp is a moot point.

Ease of driver access is an important consideration because if the driver fails before an important session or live gig, you might need to replace it quickly. The speaker is front-mounted in the new model, so accessing the speaker no longer requires the cumbersome removal of the internal speaker baffle, as it did in the first model I received. The microphone and acoustic foam must still be removed before you can reach the driver.

LOAD UP

The Micro Room's internal speaker typically uses only three to four watts of power from the source amp, so approximately 95 percent of the amp's energy is converted to heat via a resistor-based pad circuit. The 100-watt, wirewound resistor mounted in the speaker cavity takes the brunt of the amplifier output. The pad, which reduces the level by approximately 26 dB, is always in-line and has low- and high-power settings.

Resistor pads aren't the most forgiving method of reducing speaker volume, sonically speaking. Because the pads exhibit little reactance compared to a speaker, high- and low-frequency response anomalies result. In addition, the pad gives the amp a linear load that alters both performance dynamics (e.g., how the amp "feels" when the player bites into notes) and the amp's natural tone. This model contains an inductor in series with the pad and speaker to help offset the linearity of the load with some reactance from the inductor.

BLAST OFF

I tested the Micro Room during several recording sessions by splitting the output of a Gibson ES-175 guitar and running it through two guitar rigs. The "control" rig was a mid-1960s Fender Bassman head connected to a 4 × 12 sealed speaker cabinet loaded with Celestion G12M greenbacks. An identical Bassman head (a touring spare) was patched into the Micro Room, and both test rigs were routed through identical mic preamps and compressors. All signals were tracked to separate tape tracks so I could compare the sounds of the Micro Room and the conventional amplifier/microphone rig on playback.

My first test lasted through a song and a half in a poorly isolated home studio. The amp produced a volume level consistently over 100 dB, and the next-door neighbor threatened police action unless we "turned off that racket." The tracks were completed using the Micro Room, which allowed the band to play together with absolutely zero guitar bleed into the other microphones.

Although it's improved from the original model, the new Micro Room added a strong comb-filter, box-like quality to the guitar tone that I could not remove with equalization. To achieve even usable results, radical equalization was necessary. Very high frequencies, although aggressive, were controllable by rolling off 10 dB at 12 kHz and up. Fixing a huge resonance bump at 150 to 200 Hz required up to 15 dB of cut, and I had to boost the 2 kHz band by 9 to 10 dB to compensate for a serious dip. The results sounded lifeless, and the instrument's tone was certainly compromised. In the end, the sound of the Micro Room still did not come close to the thick, bell-like tone produced by the conventionally miked Bassman with no EQ applied.
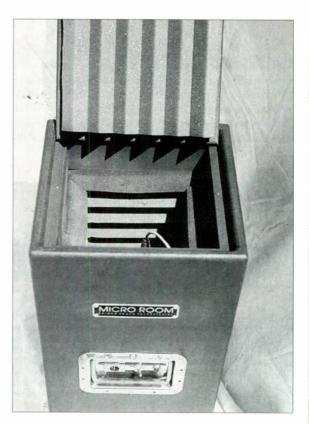
For bass, I ran an Ampeg SVT head into both the Micro Room and a stock Ampeg 8 × 10 cabinet. The Micro Room's high-end sizzle was appropriate for this application although the upper bass was somewhat exaggerated by the unit's frequency response in that region.

The original Micro Room also did a credible job for a local harp player who has an affinity for hyper-overdriven harmonica tones. Here, the fantastic isolation of the Micro Room eliminated the feedback problems I've usually encountered with this artist. In addition, the comb filtering worked well with the harp tone. The newer model, with its frequency dip at 2 kHz, sucked the presence from the harp tone.

CONCLUSION

Does the Micro Room really sound like a 4 × 12 cabinet miked with an SM57? Nope. Not even in the ballpark. In addition, finicky players may be skeptical of the "mic in a small box" design that uses a resistor pad to handle the speaker load. The small container produces

You can dramatically alter the Micro Room's sound by repositioning the microphone.

unmistakable comb-filter artifacts that are all but impossible to conceal, and the system just does not react naturally to performance dynamics.

This doesn't mean that the Micro Room is a washout. One advantage is the unit's stability: it always sounds the same. Amplifiers and cabinets never deliver identical tones from studio to studio due to ever-changing rooms, microphones, and sunspots. In addition, live players can enjoy the Micro Room's plug-in-and-play convenience and its excellent onstage isolation; when I cranked a 50-watt amp into the unit, the Micro Room produced just a faint murmur.

More importantly, although Folded Space Technologies' Micro Room does not produce an ideal 4 × 12 speaker-cabinet tone, it delivers a characteristic sound all its own. As guitarists are always searching for new, unique sounds, the Micro Room's sonic personality may be its main asset.

Richard Chycki is a producer-engineer-guitarist in Toronto, Canada, who has recently completed recordings with the Jeff Healey Band for Steven Seagal's upcoming movie Glimmerman.

Product Summary

PRODUCT:
Micro Room speaker and mic cabinet
PRICE:
$395
MANUFACTURER:
Folded Space Technologies
tel. (770) 427-8288
fax (404) 321-5094
e-mail 74602.1733@
compuserve.com

EM METERS	RATING PRODUCTS FROM 1 TO 5				
FEATURES	●	●	●		
EASE OF USE	●	●	●	●	◖
AUDIO QUALITY	●	●	◖		
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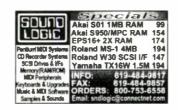
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It's Not the Pits

The dream of reusable CDs is about to be realized.

By Scott Wilkinson

Compact discs offer an excellent storage and distribution medium for audio and other types of data. However, once data is written on a commercial CD or recordable CD (CD-R), it cannot be altered. (For more information on CD-R, see "Burn, Baby, Burn!" in the June 1996 **EM**). Wouldn't it be great if you could use a CD just as you would an audio or video tape or a floppy disk?

Philips is addressing this problem with a new type of compact disc called CD-Rewritable (CD-RW). This new CD format is being supported by many companies, including IBM, Hewlett-Packard, Sony, and Ricoh. CD-RW drives will be able to read commercial CDs and read and write CD-R discs. However, current CD drives can't read CD-RW discs without a few minor modifications.

All CDs are plastic discs with a spiral groove, much like vinyl records. Unlike vinyl records, though, CDs are read from the inside out. On commercial CDs, the data is represented by a series of tiny pits in the floor of the groove. The depth of these pits is roughly one quarter of the laser's wavelength, so the light reflected from the pits is about 180° out of phase with the light reflected from the surrounding area. This phase difference causes destructive interference, which reduces the overall intensity of the reflected light to about 30 percent of the incident light's intensity. Areas without a pit reflect about 70 percent of the incident light. These two reflected intensities represent the 0s and 1s of the data.

CD-R discs include an extra recording layer of photosensitive dye molded into a blank groove (which is called a *pregroove*). When the laser writes data, the dye absorbs energy from the light and becomes less reflective in that area. The result mimics the reflective intensity of the pits in standard CDs.

CD-RW discs use a 6-layer design (see **Fig. 1**). Instead of a dye, the recording layer is an alloy of silver, indium, antimony, and tellurium. When a region of this alloy absorbs the correct amount of energy from the laser, it changes from a highly reflective crystalline state to a less reflective amorphous state.

This process requires a relatively powerful laser (8 to 14 mW). In order to return the amorphous areas to their more reflective state, a less powerful laser beam is used (4 to 8 mW). A technique called Running OPC (Optimum Power Calibration) determines the correct power values to use on each disc when it is inserted. In addition, it actively monitors the recording process and adjusts the laser's power to compensate for surface contamination, such as fingerprints, scratches, and so on.
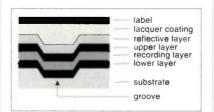

FIG. 1: The CD-RW recording layer is sandwiched between two dielectric layers that keep the heat from the laser in the recording layer and away from the substrate and reflective layers. They also fine-tune the optical response of the disc and physically hold the recording layer in place.

The crystalline and amorphous areas within the recording layer have reflectivities of about 25 and 15 percent, respectively. This is a much smaller range than conventional CDs and CD-Rs exhibit, which means that currently available CD players and CD-ROM drives can't reliably read CD-RW discs. Among the required modifications to standard drives is an automatic gain control (AGC) circuit that detects the reflectivity of a disc upon insertion and adjusts the amplification accordingly.

CD-RW is expected to make its debut by the spring of 1997 and perhaps even as soon as the end of 1996. Initially, the primary application of this new technology will be data backup. If the format catches on, we might see audio CD players and CD-ROM drives with the necessary modifications to read CD-RW discs. In this event, CD-RW could become the next mass-market medium for digital storage and distribution. Certainly, it provides what most consumers want: reusable compact discs. The compatibility of CD-RW with the upcoming digital video disc (DVD) format is another story and will have to wait until DVD becomes commercially available. In the meantime, CD-RW is another interesting step in the evolution of digital storage on optical discs. ◉

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