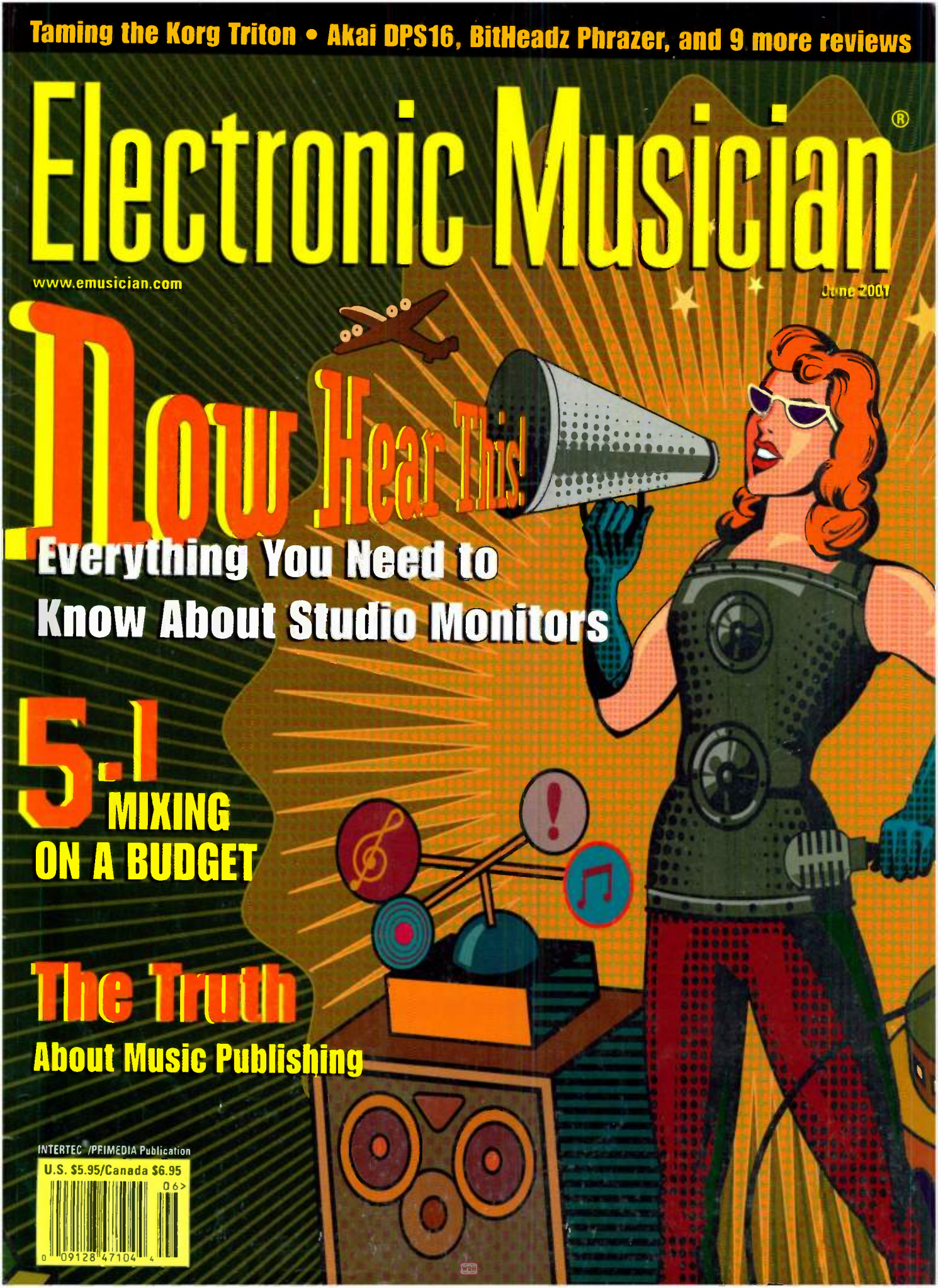
Taming the Korg Triton • Akai DPS16, BitHeadz Phrazer, and 9 more reviews

Electronic Musician®

www.emusician.com

June 2001



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RECORD! EDIT! CREATE!

Plug in the HDR24/96 Recorder/Editor and start recording. No computer to boot up. No hardware and software configuration nightmares. No compromises like settling for 20-bit audio or just eight tracks at a time.

Recording's easy with the HDR24/96.

Simultaneously record twenty-four tracks of 24-bit digital audio...without waiting for lock-up, tape shuttle or CPU lag. Drop up to 192 alternate takes into "virtual tracks." Record onto affordable, removable media that you can swap in and out for each project.

And do it all with your hands on a familiar, analog-style machine (or choose from two sizes of wired remotes) instead of resorting to myriad mouse clicks. All basic functions are right on the HDR24/96 front panel including transport buttons and a Record Enable button for each track.

Editing is easy with the HDR24/96.

Plug in an SVGA monitor, keyboard and mouse, choose from 2x, 4x, 8x, 12x or 24-track views and then watch them scroll smoothly past a centerline.

Mark hundreds of cue points and four locate points for looping and auto-punch-in modes.

Use the mouse to "scrub" individual tracks, Cue, Punch and Loop points with continuously variable velocity.

You can mark a segment (or multiple non-adjacent segments) as a *region* and then cut, copy and paste it anywhere — onto a blank track or right in the middle of an existing track without erasing anything (the part of the track after the insert just "slides down").

You can audition regions or modify their start/end points instantly, capture them as "sound elements" for later use or quantize them to user-defined time grids.

Create fade-ins, fade-outs and cross-fades just by dragging and dropping them ...and then set their length by dragging the mouse.

Add volume envelopes for simple level automation of regions or whole tracks.

Then use Render Track to combine all or selected regions of a track just as you hear it complete with cross-fades, volume envelopes, mutes, etc.

Play with the HDR24/96.

Play back twenty-four tracks of pristine digital audio —instantly without any pause or lag time. It will be synched rock-solidly to everything in your studio — from MIDI-based sequencers to VTRs (via SMPTE or video sync).

If only life had 999 levels of undo. HDR24/96's History list lets you take loads of creative chances.

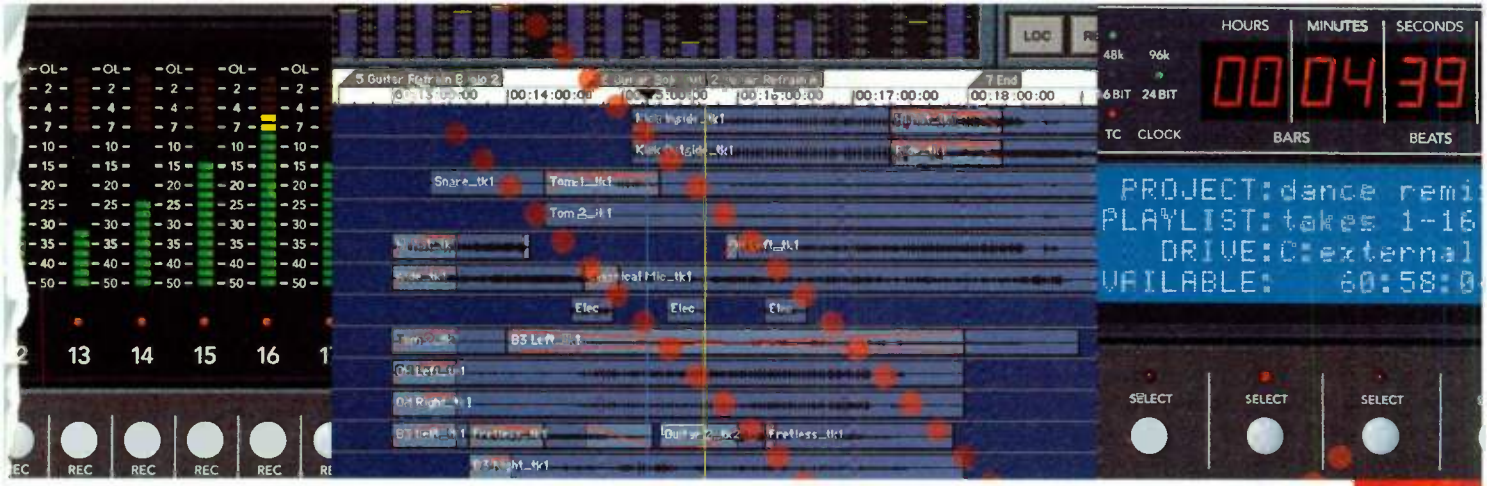
Then let your partners, clients and friends "play" with your tracks anywhere in the world, thanks to the HDR24/96's Ethernet port and FTP server capability.

The non-linear HDR24/96 vs. linear hard disk recorders.

Ever since the invention of magnetic tape, recording over something means it's

"...the HDR24/96 is a stunning development with excellent sonic quality, an extensive feature set and versatile file management... it's easy to use and priced right. This one rocks!"

George Petersen
Mix Magazine March 2001



SAVE YOUR COMPUTER FOR E-MAIL.

Need to back up just one song? Plug a Mackie Media Project drive into the HDR24/96 external bay and transfer over 2GB to an ORB™ disk.

gone…which makes doing "punch-ins" a dicey gamble. This is called *linear* (destructive) recording. Even some current hard disk recorders use this old-fashioned technology!

The HD24/96 employs true, non-destructive, *non-linear* recording and editing. That means you can record as many versions of a track or track segment as you want without destroying the original. During playback, the recorder recombines the non-linear segments into a seamless soundstream.

Professional remote for a very professional hard disk recorder. Our new Remote 48 lets you run two HDR24/96s – 48 tracks of total control including a weighted jog/shuttle wheel and full display!

And unlike linear-style recorders that treat disk space like digital tape, the HDR24/96 doesn't automatically eat up 24 tracks of disk space when you're just recording one or two tracks. Because it uses only the space needed for actual audio, you get far more recording time per gigabyte of hard disk space.

Professional performance *and* affordable creativity with the HDR24/96.

Non-linear hard disk recording *is* possible to do with a computer-based system. But to achieve what the HDR24/96 delivers – simultaneous, lag-free 24-track/24-bit recording and playback and waveform accurate editing – requires major investment in a *very* expensive digital audio workstation system. Cheap "recorders-on-a-computer card" just don't have the horsepower for multi-track, twenty-four-bit 48kHz recording, much less *twelve-channel 96kHz capability* like the HDR24/96.

Listen to somebody else instead of us.

Here's what ***Mix*** magazine had to say about the HDR24/96:

"…The HDR24/96 is a stunning development with excellent sonic quality…The unit offers an ease of use that should make disk-recording novices comfortable while including an impressive feature set that will appeal to seasoned pros.

"The recorder's faceplate holds few mysteries and most users can be up and recording just minutes after unpacking the HDR24/96."

According to Britain's ***Audio Media***, *"As a recorder (the HDR24/96) is transparent. As a tool, it's powerful. As a creative helper it's perfect. With focus on functional, inexpensive, simple-to-use 24-track recording, Mackie has hit the mark."*

Get a demo at a Mackie dealer.

There are a bewildering array of digital recording options on the market right now. You've heard our two cents worth.

We honestly believe that we've created the best of two worlds: the best standalone non-linear digital recorder, and an extremely robust editing system with ultra-functional graphic user interface. And we've done it without making you enter the *really* cruel world of computer interface compatibility problems.

Twenty-four track masters for under ten bucks each!! Divide the cost of a MackieMedia M90 into the 20+ pop tunes you can record on it and you're looking at under a ten-spot for each 24-track master. Remember, non-linear hard drives store audio data only, not silence. Tape (and linear hard disk recorders) just roll merrily along…eating oxide and costing money.*

Call toll-free or visit our web site (using that computer you won't need to tie up) for more info.

Then get your hands on an HDR24/96 and track some hits.

www.mackie.com • 800.258.6883

Get the MX-2424 Advantage!

"The 'Golden Ears' all found the TASCAM performed extremely well, nearly beating out a unit priced four times higher. The listening tests confirmed what I already knew: the MX-2424 is a solid performer at a great price."

- Glen O'Hara,
Pro Audio Review Magazine

"...the MX-2424 puts high-resolution sound quality and professional recording features at your fingertips."

- Electronic Musician Magazine,
2001 Editor's Choice Award

"...the TASCAM MX-2424 is a rock-solid, excellent studio recorder that performs well, sounds great and is priced right."

- George Petersen,
Mix Magazine

"The machine alone is impressive enough to warrant close attention, but the implications inherent in the control and networking capabilities make it potentially astounding."

- Rob James,
Studio Sound Magazine

You probably
machines in
is the
What you
advantages
multitracks,
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just getting
understand

MX-2424 24-TRACK 24-BIT HARD DISK RECORDER/EDITOR

* based on an average 3 1/2 minute song of 24 tracks at 24-bit/48kHz. Your mileage may vary.

† Offline CD-R backup is possible with an Ethernet-equipped computer. The $749 (USD) reference is based on TASCAM's CDR-Pro Bundle.

know that with thousands and thousands of use around the world, the **TASCAM MX-2424** most popular 24-track recorder ever made. might not know is that the MX-2424 offers huge that aren't available on other standalone hard disk regardless of price. Whether you're making the from analog and tape-based digital recorders or into recording, here's some info to help you truly the MX-2424 advantage.

Ph.D. in Nuclear Physics Not Required

If you've ever recorded before, you'll find the MX-2424 as easy to use as any multitrack recorder. Flip the Power switch, arm a track and hit the Record and Play buttons. Voila...you're tracking to its internal hard disk. Since TASCAM has been the world leader in multitrack recording for over 25 years, we know how to create gear that's powerful and sophisticated without making the learning curve too steep.

Edit How *You* Like: MX-View™ Waveform Graphic Interface and Extensive Front Panel Editing

One of the main reasons to get into hard disk recording is the incredible editing power versus tape. Running in native Mac and PC versions and connected via a fast 100Mb Ethernet interface to your computer, the upcoming MX-View is a powerful graphic editing interface that offers sophisticated, sample-level editing on par with full-featured digital audio workstations. You can drag and drop on the fly, get onscreen metering for up to six MX-2424s, set up custom configurable keyboard shortcuts, manage virtual tracks and much more. If you want to use the MX-2424 in the field, its extensive built-in front panel editing tools let you edit without lugging around a keyboard, monitor and mouse.

True Recording Power: Take the Punch-In Challenge

24-track, 24-bit digital audio requires a powerful hard disk recording engine. The MX-2424 is so strong that it allows for seamless, gapless punches across 24 tracks, with up to 72 tracks of throughput to accomplish this considerable task. If you're brave, try arming 24 tracks on any other standalone 24-track hard disk recorder and quickly punching in and out. It's just one example of the MX-2424's awesome dual-processor recording power and extremely fast SCSI bus. You can choose between TapeMode and Non Destructive recording, and access up to 999 virtual tracks per project with 100 locate points, 100 levels of Undo and much more.

Sound Designer II, Broadcast Wave Files and SCSI Drives for Ultra Flexible Compatibility

TASCAM understands the reality that you may need to interface your audio with other pieces of equipment. Since the MX-2424 writes Sound Designer II™ audio files to Mac-formatted disks and Broadcast Wave audio files to PC disks, it's easy to move sound back and forth between your computer and the MX-2424. With these standard time-stamped file types and professional SCSI drives, you're ensured sample-accurate compatibility with Pro Tools™, Nuendo™, Digital Performer™ and more. With compatibility being so important to MX-2424 owners, it's no surprise that its 24-channel interfaces are ready to connect to just about any console, digital or analog. Or that its analog, TDIF and AES/EBU interface modules are 96kHz ready.

Back Up Your Tracks: As Low As A Buck Per Song

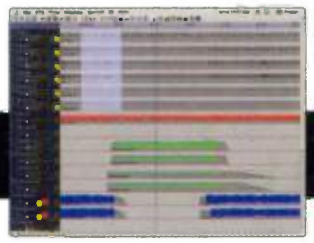
Media	Cost of Drive	Media/10 Projects	Total Cost
90 Minute IDE Drive	$299	10 Drives	$2990
Orb Drive	$299	1 Drive + 86 Disks	$2879
TASCAM DVD-RAM	$599	1 Drive + 20 Disks	$1739
Offline CD-R Backup	$749	1 Drive + 290 Disks	$959

If you're forced to use cheap disk drives to backup, you'll pay in the long run. DVD-RAM drives may be connected to the MX-2424's front panel or rear SCSI port, and offline CD-R backup via Ethernet transfer to your computer is the most cost-effective backup method available on any HD recorder by far.

Hard disks are great for recording...but not so great for archiving and transferring audio. That's why the MX-2424 gives you choices like 9.4GB DVD-RAM discs for your backup solution. Or simply transfer your audio to your computer and backup to CD-ROM for as low as one dollar for an average pop tune*.

Get the Advantage of the Most Powerful and Most Affordable 24-Track Hard Disk Recorder Available Today

There's much more to the MX-2424 than what fits on this page, like its award-winning sound quality, professional built-in synchronization tools and TASCAM's amazing online support forums. So if you're getting into the hard disk revolution, you might as well take advantage of the recorder with all the advantages. Just go to **www.mx2424.com** for the complete MX-2424 story, or check out the MX-2424 for yourself at any TASCAM dealer.



Available soon, the new MX-View graphic editing software offers DAW-style waveform editing power, drag-and-drop editing on the fly, control of up to six MX-2424s with metering and much more.

www.mx2424.com

TASCAM
a whole world of recording

I N S I

FEATURES

DEPARTMENTS

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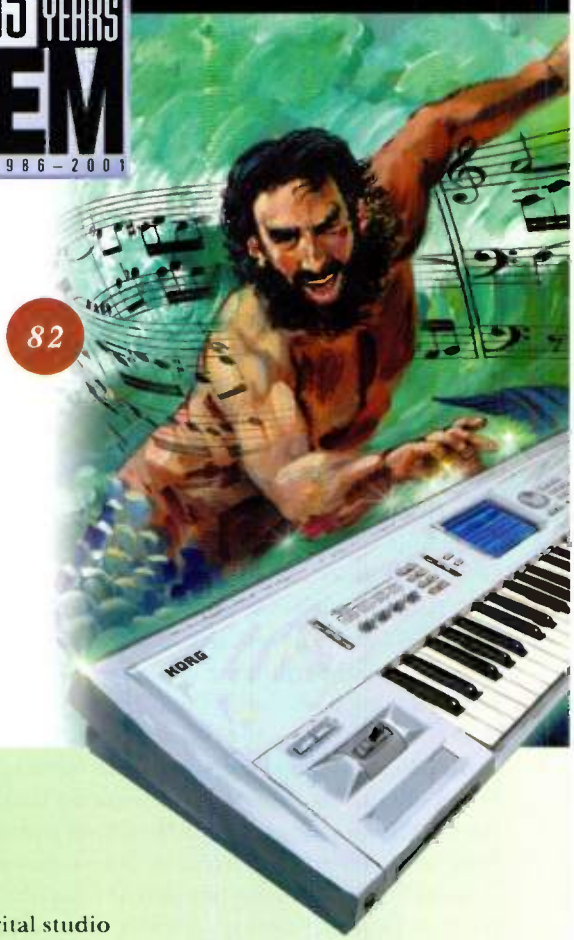

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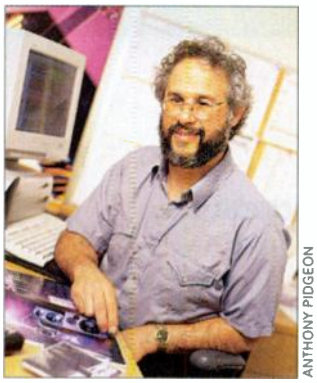

ANTHONY PIDGEON

In the past six months, we have covered a lot of new ground. In addition to our usual in-depth stories about music technology, we have expanded our coverage of music production for the Web and explored the world of 5.1-surround mixing. We've launched *Remix* and *Onstage* as monthly magazines, and our Web site has undergone a radical redesign. This midyear editorial column seems like a good place to pause and reflect.

For the most part, I think we've done well. Our coverage of music production for the Web has been well received, and we plan to do a lot more. Surround mixing is still somewhat bleeding edge for personal-studio owners, but musicians need to stay informed about this important development, so we're going to keep forging ahead.

I am delighted with *Remix*, which has delivered its promise of providing a quality technical magazine for underground music production and DJ performance. *Remix* editor Chris Gill has done a superb job of keeping his finger on the pulse of the electronic-music world.

I'm also proud of the work *Onstage* editors Mike Levine, Barry Cleveland, and Chris Kelsey have done. They've delivered quality interviews with major artists, emphasizing ideas performing musicians can use. In addition, the editors provide accurate hands-on reviews and practical technical articles for musicians who play live.

The Web site redesign has been another matter. My sincere thanks to those who have written to us offering suggestions. Your input is helpful and is being taken seriously. We're still striving to make www.emusician.com what we want it to be, including addressing issues such as the article archives and audio and video example files. Hopefully, by the time you read this, we'll have started making the necessary changes.

Speaking of Web sites, our parent company, Intertec, has added a new feature to all of its sites. Starting with this issue, you will see as many as three items in "What's New" marked with a new icon that says "Demo Room." That indicates that a manufacturer-supplied demo video for the product is available in the new Demo Room section of www.emusician.com. The number of Demo Room products will vary from issue to issue, but we hope to offer at least one each month. Manufacturers can also purchase space in the Demo Room to show videos. Those paid videos are essentially advertisements and are not reviewed by the editors; the editors only select the Demo Room videos mentioned in "What's New."

We've also added something new to the print magazine. Starting with this issue, any product reviewed in "Quick Picks" that earns a perfect score of 5 in the Overall rating will be awarded our new Hot Picks icon, which will be placed at the end of the review. We are chary about awarding perfect scores, so when you see a Hot Picks icon, you know we have found the product to be truly exceptional.

I have one other change to announce. Unfortunately, long-time technical editor (and my close friend) Scott Wilkinson is moving on to greener pastures. Scott has contributed immeasurably to **EM**'s success, and we'll miss him.

We welcome your feedback. E-mail us at emeditorial@intertec.com

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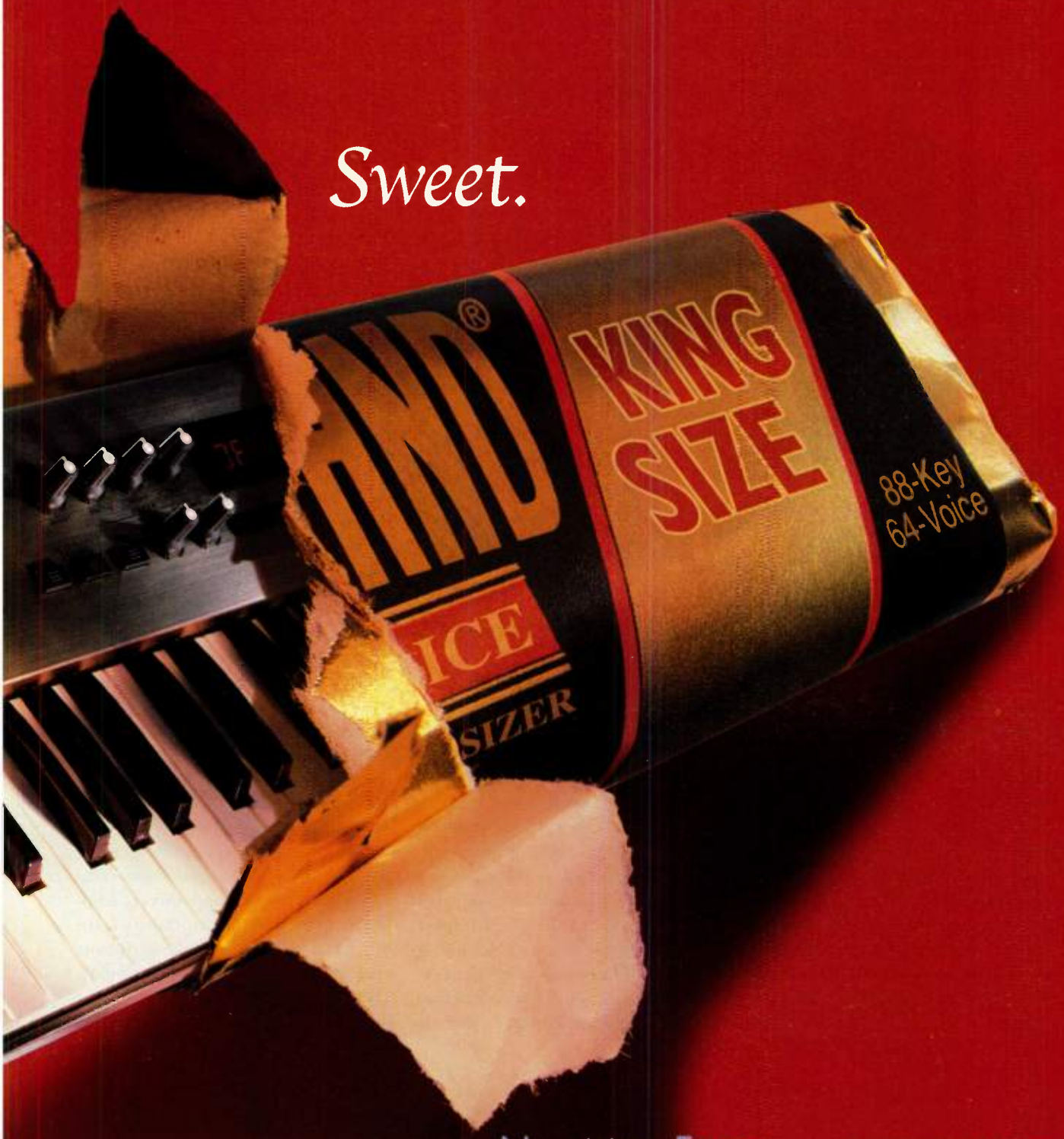

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MIKE CRUZ

OPPRESSED MALES UNITE!

I'm curious when you will be running the companion article to "Electric Ladyland" (April 2001) called "Electric Guyland."

When you plan articles such as "Electric Ladyland," do you give any thought to the impression you might leave with your subscribers? If you were to publish the article I alluded to, I bet the fallout about male chauvinism would be prodigious and extreme. Why is reverse discrimination acceptable?

An article such as "Electric Ladyland," though perhaps well intentioned and full of meaningful insights and information, is sexist. I am sure the article could have been presented in a more considerate way.

Gene Lowinger
via e-mail

NO DISRESPECT TAKEN

I am a producer (hardware and software) and also a DJ. I completely agree with Thomas Sobcak's comments ("Letters," April 2001) about DJ Rap and her hypocritical judgments of Madonna.

I was delighted to see the interviews with women musicians and engineers ("Breaking the Sound Barrier," February 2001). Although **EM** does an excellent job covering gear, software, and many other things, it should include more interviews with artists and DJs. Not to dis the magazine, but although the editors may be the true experts, people like to hear from the big names and the celebrities. Perhaps **EM** could add a new section for interviews; comments; tidbits of information; and even discussions about the artists' gear, skills, and so on.

Sean Angelo
via e-mail

WON'T GET FOOLED AGAIN?

I'm intrigued by the "Tech Page" article, "Xena: Wired Songstress," in the April 2001 issue. It mentions a new product that sounds exciting, but I have to ask: is that for real? Lucy Lawful and Gaby Realle? Melissa Ethernet? As much as I would like to, I have a hard time taking them seriously! I thought I was perhaps reading *Games* and had found the fake ad. I also tried to go to the Internet site listed in the article, www.xenaxing.com, but all I saw was an error page. Can you give me any information that might validate the company and its product?

Dee Adams
via e-mail

Dee—April Fools'!—Scott Wilkinson

GEAR, SCHMEAR

The April 2001 issue's "Mysteries of Mixing" was most interesting and needed. Much too often, gear becomes the godhead; whereas, the ears and the song get pushed to the back of the mix. Chris Lord-Alge's point about a lot of engineers thinking the mix is about gear—when it's more about gut instinct—should be written in gold and put in every studio. Keep up the great work.

Burt Teague
East Granby, CT

Many of us in the music world are professional musicians and writers who stumbled into recording through the unlocked back door. We began using the simple tools of yesteryear to write or lay down tracks while sitting in the comfort of home. Then those simple pieces were brought to the "real" studio, where the recording, mixing, and mastering were done by trained professionals. Now, years later, many people have home and project studios with equipment that far surpasses that of the "big boys" of days gone by (I'm still amazed when I sit in front of my Digidesign Pro Tools MixPlus system, with its bank of shimmering vintage compressors).

As home-studio equipment grew in power and finesse, we honed our skills as pseudorecording and -mixing engineers; yet trained technicians we are not. We remain primarily musicians and writers. Short of attending an audio-recording school, one fix for those technical shortcomings is voracious reading. Unfortunately, that reading too often leaves me disappointed. Too many articles begin with promising titles and leave me thinking, "So what?"

"Mysteries of Mixing," in the April issue, was quite different. Simply put, the article was fabulous and informative. Of all the articles in publications that I receive on a monthly basis—and generally read from cover to cover—that piece was the best I've read in years. I, for one, advocate more stories of that type.

Felix Misch
via e-mail

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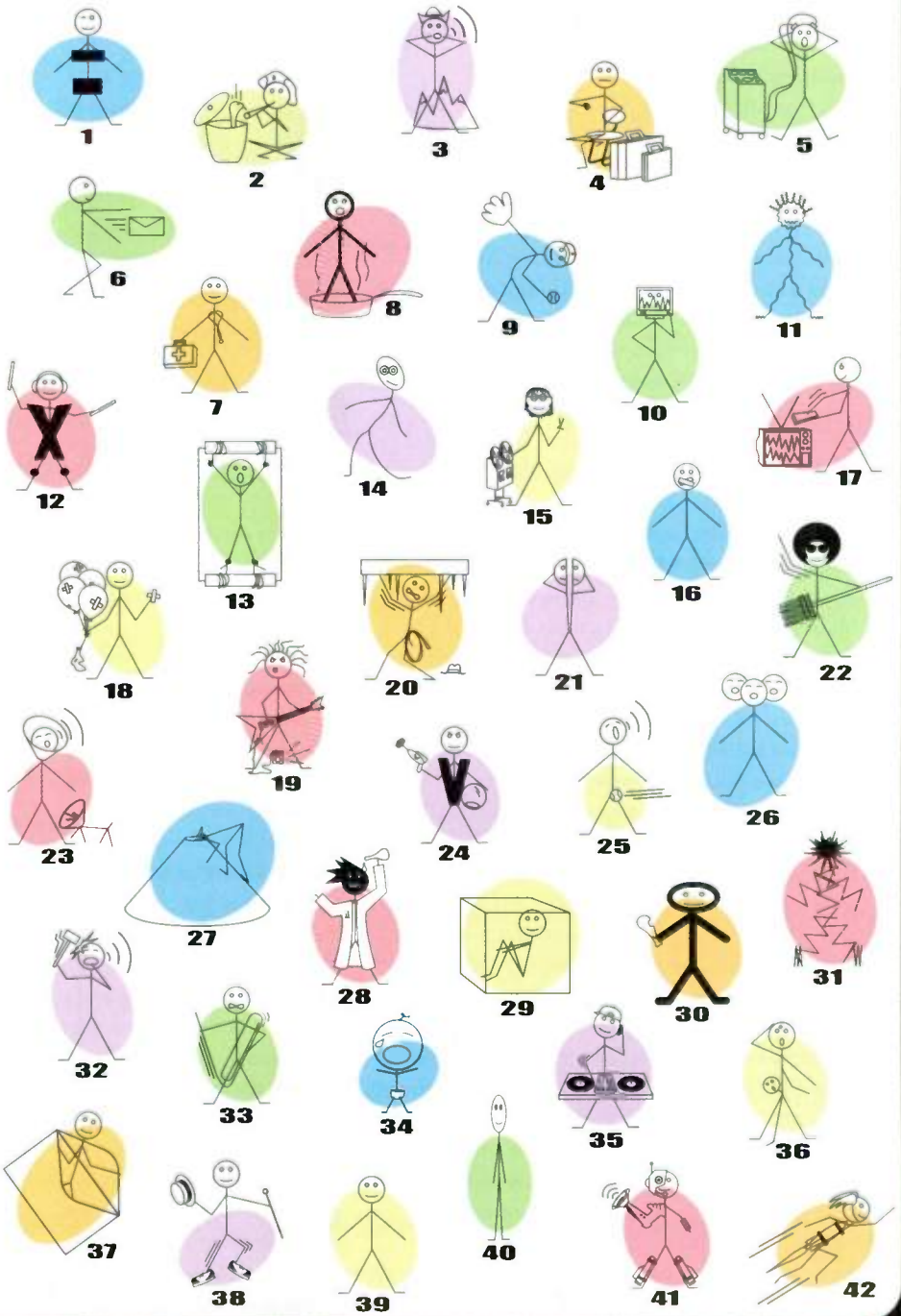

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Win a Cool Edit Pro Stickmen T-Shirt!!

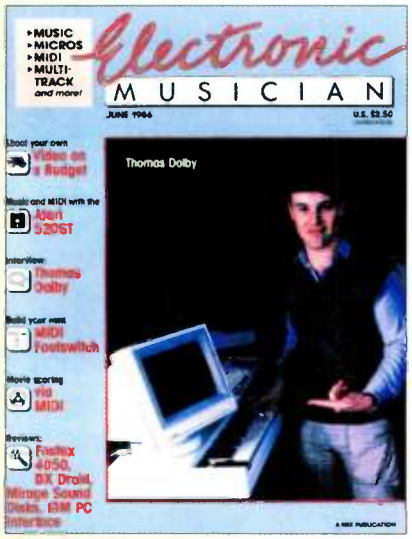
Match each stick figure to the corresponding feature name on the left. Visit www.cooledit.com/sticks to view the answers and to enter a drawing for one of twenty-five free stickmen T-shirts! All entries must be submitted by July 15, 2001.

As always, no stick figures were hurt in the making of this ad- well, except for maybe #25.

WRH

FIFTEEN YEARS AGO IN EM

For a music-technology magazine, we were obsessed with video and film in some of our early issues. The June 1986 issue included no less than four video-oriented stories and one film-oriented article. Most of those articles involved electronic music, but not all.



We started with Jack Orman's story on how to take advantage of local public-access cable television. David Carr offered a step-by-step tutorial on shooting a music video. Don Slepian left the music world behind in his story on video synthesis, and Orman provided a DIY project on building a simple video-distribution amp. Finally, Robert Kraft brought music back to the fore by discussing how to score a film, using MIDI gear.

Our computers and software section focused heavily on the Atari 520ST, the first computer with a built-in MIDI interface. Malcolm Cecil introduced the 520ST, which was based on the same 8 MHz Motorola 68000 processor found in the Macintosh and Amiga computers of the day. The Atari machine was fast for its time and featured a decent onboard sound generator. Its clock timing was more accurate than that of a Mac, PC, or Amiga, so a well-written sequencer for the ST could be impressively precise. All that made it a popular computer for musicians, especially in Europe. To get us started, Eric Barbour demonstrated how to program a MIDI sequencer for the ST.

Our June issue had two other DIYs. Tim Dowty showed us how to build a MIDI Program Change footswitch, which was perfect for guitarists who wanted to control their effects onstage. Thomas Henry offered an assortment of small modules you could add to a modular analog synth to fill the empty panel slots, including a multiple (signal splitter) and a pulse extractor that recovered timing pulses from a taped clock track.

We returned to the Atari 520ST in our reviews section with **EM** founding editor Craig Anderton's evaluation of Hybrid Arts' *DX Droid* editing software for the ever-popular Yamaha DX/TX–series FM synths. Patrick Hubbard checked out the Fostex 4050 SMPTE/MIDI synchronizer, and Paul Grupp reviewed the Intelligent Music OP-4001 MIDI interface for PC-compatibles.

The June 1986 issue also featured one of the most questionable decisions in our 15-year history: David Doty reviewed Anderton's book about digital delays, *The DDL Handbook*. Doty was a qualified and honest reviewer, and Anderton gave him free rein. Nevertheless, in my view, it was unethical, and we shouldn't have done it.

I saved the best for last: our cover story about a beret-clad guy you may have heard of who recorded a piece called "She Blinded Me with Science," scored *Howard the Duck* for LucasFilm, collaborated with George Clinton, produced a Joni Mitchell album, and generally worked on the cutting edge. Even then, Thomas Dolby was a fascinating man, and Anderton obviously enjoyed interviewing him. They discussed Dolby's gear, his approach to composition, his various collaborations, how he structured his recording sessions, and much more.

—Steve Oppenheimer

● LETTERS

SQUEEZE PLAY

I just reread the cover story about compressors ("The Big Squeeze," February 2001), and I want to thank you for Michael Cooper's excellent article. The story coincided with my decision to purchase a compressor for my home studio. Cooper's article laid everything out in a straightforward manner, and it really helped me with my purchase.

Oddly enough, the compressor that I chose was not even mentioned in the article. FMR Audio's RNC-1173 (RNC stands for "Really Nice Compressor") has been described as the "best compressor for less than $1,000." Not bad, considering that the unit costs less than $200.

I've been amazed at the quality of the unit—it's very transparent. True, you don't have balanced ins and outs, but it's still pretty impressive. I recommend it to anyone putting together a home studio.

Keith Hanlon
Columbus, OH

KEITH HAS ME CLAV

Julian Colbeck's March 2001 "Vintage Page" about the Hohner Clavinet D6 included not only some very useful bits of information but also some inaccuracies.

First, the Clavinet retailed for $1,195, not $700—I know that because I sold them in the '70s. I sold one to Keith Richards.

Second, the problem with the pads 95 percent of the time is the groove that wears in them from the strings, which inadvertently causes the plucking sound when the key is released. Some people find that desirable. For those who don't, simply loosen the fitting that holds the pad and rotate the pad 90 degrees so that its groove goes against the string angle.

Good string replacements are loop-end banjo strings. You might have to experiment with the gauge, but get the brightest possible. Burn the synthetic dampening material off the string's loop end and wrap the loop around

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the anchor on the harp assembly. Don't disturb the yarn woven through the strings. If you do, you will get varying decay times when the keys are depressed.

Colbeck writes that the Clavinet "requires a 9V battery." Wrong! It does require 9V but not in a battery format. The newer D6 units shipped with a 9V wall wart.

Aside from the effects mentioned in the article, distortion boxes are effective if you bear in mind that the Clavinet is nothing more than a big guitar laid on its back with a keyboard over it.

Keith "Plex" Barnhart
via e-mail

Keith—Thanks for your endorsement of at least some of the material contained in the Clavinet article.

I think you may be confusing the D6 with the D7 (hey, the '70s were an awful long time ago), which came out at the end of the decade and was indeed priced substantially higher than the D6. That said, I doubt Keith Richards would have noticed if you charged him ten grand for a pack of cigarettes back then.

The D7 came with the wall wart you mention, which again leads me to think you might be confusing the two models. Just to make things really tedious, some early D7 models were labeled "D6."

Your comments about pad wear and tear and string tone are interesting. Unfortunately, because of space constraints, "Vintage Page" can offer only a bird's-eye view of each subject. I attempt to stick to information that will whet the appetite of the casual observer, rather than to provide in-depth information for experts like yourself.—Julian Colbeck

WINDOWS ME OR 2000?

I'm looking for guidance about which Windows operating system is the best choice for music production. Has anything been printed recently or is anything coming out soon that indicates whether ME or 2000 is the better choice?

Brian Strines
via e-mail

Brian—If you're just starting out or if you're thinking of upgrading from Windows 98 or earlier, put your money on Windows 2000. It is, by most accounts, the most stable and forward-looking PC operating system in the Microsoft domain. I've upgraded several of my computers to 2000 and have not had a single crash in many weeks, which is truly remarkable in the Windows world. However, I did need to get updated drivers for some of my peripherals and software upgrades for several of my nonmusic applications.

Many manufacturers, including Cakewalk and Steinberg, offer products that are highly optimized for 2000. A large number of audio-card makers provide drivers that can take advantage of what Windows 2000 offers. (Nevertheless, I know of at least one multichannel-card maker that's not quite there yet.) The operating system has significant advantages for audio and somewhat less so for MIDI use. Unlike 98 or ME, 2000 supports dual processors.

Without a doubt, Windows 2000 is your best bet for music production. Contact manufacturers to make sure their products are fully compatible with Windows 2000.—Dennis Miller

ERROR LOG

April 2001, "Vintage Page: Linn Electronics LinnDrum," p. 50: The unit pictured is a Linn LM-1, not a LinnDrum. Also, the photo should have been credited to Mickey T of the Drum Machine Museum (www.drummachine.com).

WE WELCOME YOUR FEEDBACK.

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WHAT'S NEW

By Marty Cutler

OBERHEIM OB-TUNE

Oberheim Electronics' first software plug-in is *OB-Tune* (Win; $49.99), a DirectX pitch-correction processor based on digital signal processing (DSP) technology licensed from Antares Audio Technologies. *OB-Tune* offers real-time pitch correction of monophonic source material without artifacts or distortion.

OB-Tune has an easy user interface. You get three virtual buttons: Bypass, Edit Scale, and Vibrato Type; knobs for Scale Detune, Detune Speed, Tracking Sensitivity, and vibrato Depth, Rate, and Delay; and sliders for assigning a key and scale selection. An onscreen meter shows the amount of pitch change. The plug-in has a 26-scale collection, including ethnic and microtonal scales.

OB-Tune needs a DirectX-compatible host program such as Cakewalk's *Sonar,* Steinberg's *Cubase VST* and *Wavelab,* Emagic's *Logic,* or Sonic Foundry's *Acid.* MusicYo.com (distributor); Web www.musicyo.com.

NATIVE INSTRUMENTS FM7

Yamaha's DX7 and other FM synthesizers are now re-created in software with *FM7* ($299) from Native Instruments. However, *FM7* offers features that go well beyond its hardware counterparts, including distortion and filters, effects processing, and an audio input.

Unlike hardware FM synths, which offer only preset algorithms, *FM7* comes with eight operators and lets you create operator routings.

FM7 includes a variety of waveforms other than sine waves, including sawtooth, square, and waveforms with multiple resonant peaks. Audio input can serve as an additional carrier or modulator. The distortion operator also offers a noise waveform. With unlimited stages and looping capabilities, *FM7*'s envelope generators greatly exceed those of its hardware siblings.

The synth reads System Exclusive files from DX- and TX-synths, including the four-operator varieties. For those with an interest in creating original sounds, *FM7* easily outdistances the DX7's user interface with parameters that are clearly visible on the front-panel, including a breakpoint-style envelope-generator display, operator on-off buttons, frequency-ratio displays, and even a spectroscope and oscilloscope. You can even assign MIDI continuous controllers to modulate parameters in real time.

Windows users will need a Pentium III/450 MHz computer with 64 MB RAM running Windows 98 or later. For Macs, the program requires a Power Mac G4/400 MHz running Mac OS 8.6 and 64 MB RAM. Steinberg North America (distributor); tel. (818) 678-5100; e-mail infoUSA@native-instruments.com; Web www.native-instruments.com.

ABLETON AG LIVE

Ableton AG's *Live* ($350) is a digital-audio sequencer optimized for real-time performance. *Live* launches audio clips using MIDI commands as it plays a sequence. The audio clips pool can be loops, one-shot samples, or songs. Audio streams directly from disk, avoiding impositions placed by RAM.

Live's Time-Warping Engine automatically performs time-compression and expansion chores on the fly to ensure synching of audio tracks. Tracks can be synched to *Live*'s internal clock or an external source, including MIDI Clock or MIDI Time Code. The track's pitch is independently adjustable and is not affected by time compression.

The program has a large number of VST effects plug-ins. Effects are controlled with an onscreen two-dimensional display or with MIDI messages from a remote device. *Live* records controller moves, and changes can be tweaked in the graphical display with breakpoint envelopes. *Live*'s unlimited undo history lets you revert to earlier edits. Mac users need a Power Mac 604e/200 MHz, 64 MB RAM, Mac OS 8.6, and OMS 2.38. Windows users need a Pentium/166 MHz with 64 MB RAM and Windows 95, NT 4.0, or 2000. Ableton AG; tel. 49 30 288 7630; Web www.ableton.com.

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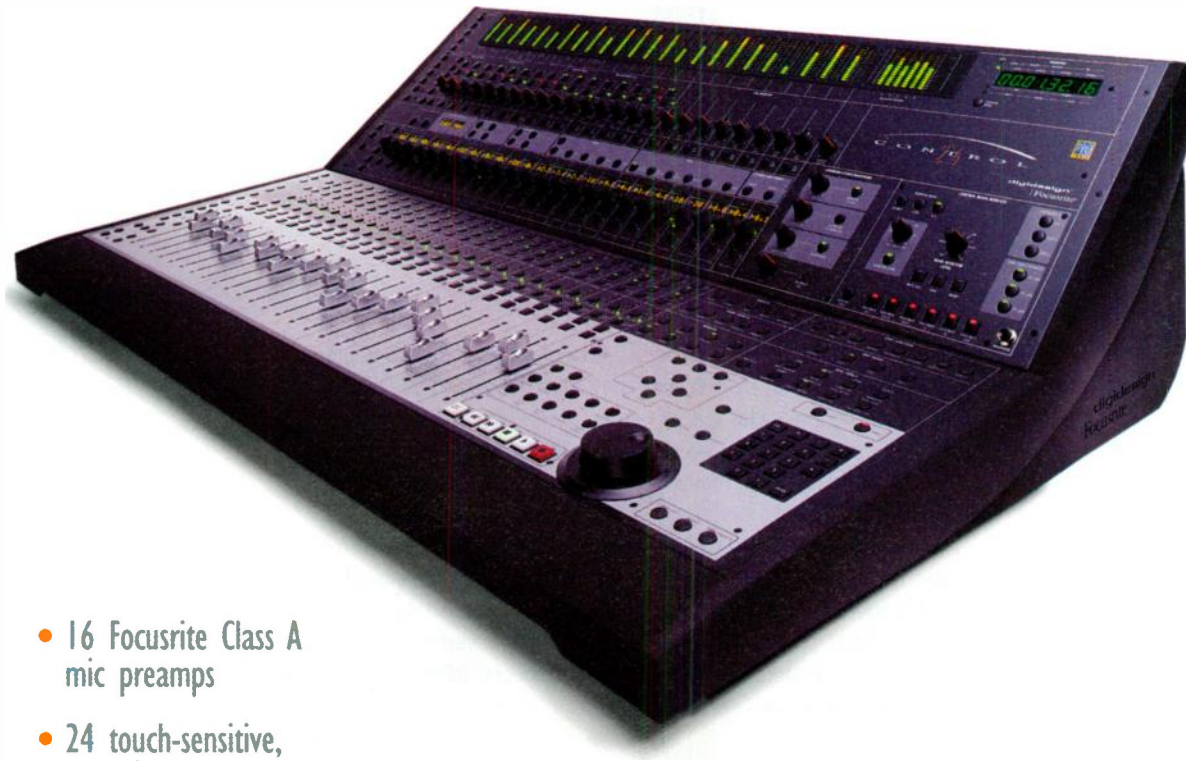

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The D1600 comes with a massive 20 GB hard drive for more recording time and less worry about running out of time or space. And thanks to its unique user-swappable design you can easily change drives from session to session using standard IDE mechanisms. The D1600 supports the largest drives possible, so say goodbye to making backups and clearing your drive for the next project. Simply swap it!

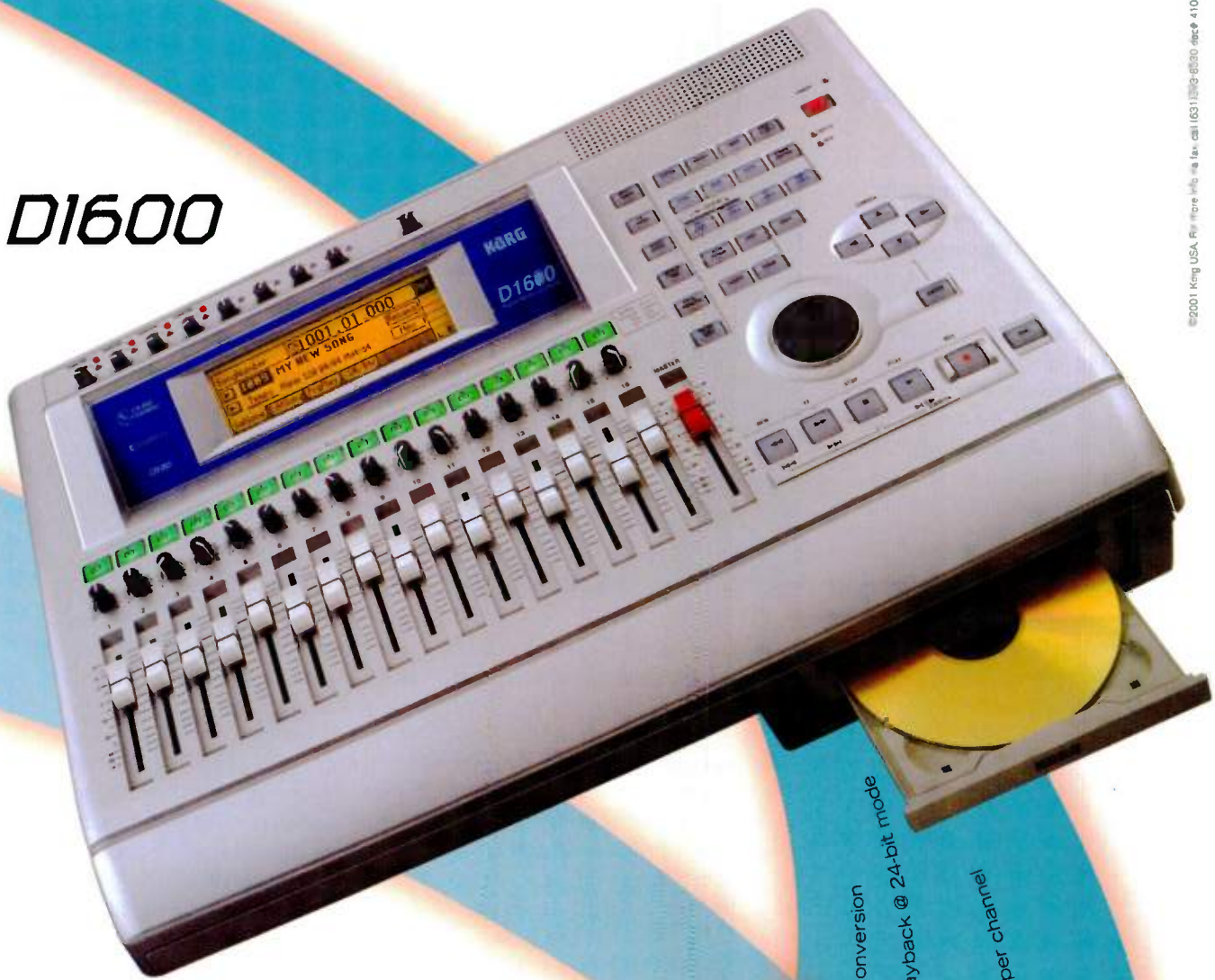
When it comes time to master or backup a project to CD you'll appreciate the D1600's internal CD-RW drive bay. (The Korg model CDRW-2 and many ATAPI-compatible devices can be used.) No cables. No additional power supplies. You can even record audio directly from the internal CD-RW drive. Try doing that with an external unit!

The effects power of the D1600 really shines when recording and mixing. Have up to eight Insert effects configured any way you like, plus two Master and one Final effect. It's like having a professional rack of high-quality effects processors with everything from reverbs and delays to compressors, limiters and EQs. Plus, our special REMS™ models of mics, guitar amps and speaker cabinets.

With its user-friendly TouchView graphic display and intuitive operating system, the D1600 is just begging to be touched. Lay your hands on it and let your creativity take over. Once you heat things up, there's no telling what you'll be able to burn.

KORG

D1600



- Recording Format: 24-bit/16-bit uncompressed, 44.1 kHz
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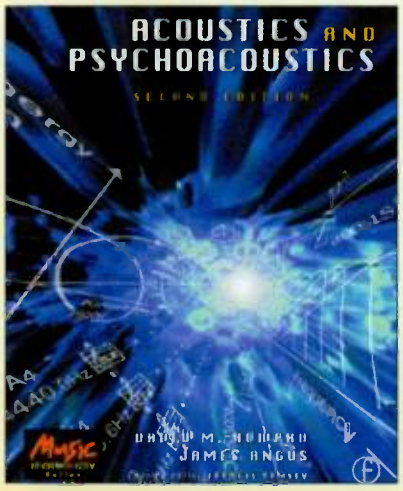
FOCAL PRESS

Acoustics and Psychoacoustics ($44.95) by David Howard and James Angus is an introduction to human sound perception. The topics covered include principles of sound; musical timbre, pitch, and amplitude; musical instrument sound propagation; and the effect of acoustic environments on sound. The book also provides information about waveform motion, brass and woodwinds, and forward and backward masking.

Focal Press maintains a Web site in support of the book, designed by coauthor Howard. The site includes questions and exercises to test knowledge, Web links for further research, audio clips, and online applications. Focal Press (Butterworth Heinemann); tel. (781) 904-2500; Web www.focalpress.com.



MUSKA & LIPMAN

Scott Garrigus's *Sound Forge Power* ($29.95) is an exhaustive resource for users of Sonic Foundry's *Sound Forge* and *Sound Forge XP* digital-audio editing software. The book's subject matter covers working with audio-file formats, editing, processing, sampling, and producing audio and video for Web distribution. Sonic Foundry edited the book for technical accuracy.

The guide has numerous screen shots illustrating examples and tutorial exercises. Garrigus offers tips, notes, and warnings about irreversible actions, which are highlighted with special icons.

Garrigus maintains a Web site (www.garrigus.com) that supports the book. The site has a discussion area for posting questions and a chat area for communicating with the author. It also offers a free monthly newsletter featuring tips and techniques that are not found in the book. Muska & Lipman; tel. 513-924-9300; e-mail publisher@muskalipman.com; Web www.muskalipman.com.

HELPFUL MUSIC VIDEOS

Analogue Synthesis Actual Reality ($35) is a 58-minute introduction to modular analog synthesizers. The video offers a generalized tour of analog synth components, including voltage-controlled oscillators (VCOs), voltage-controlled amplifiers (VCAs), envelope generators (EGs), filters, and more unusual modules. Analogue Systems modules are used for the demonstration.

The video steers away from most jargon in an attempt to dispel the mystery surrounding analog synthesis.

Topics include voltage control, patch creation, and MIDI control of an analog synth. The video ends with a tour of Museum Studios and its variety of classic modular synths, including instruments by Arp and Roland. Enport (distributor); tel. (402) 398-0198; e-mail enport@home.com; Web www .en-port.com.

NO WALLS

Music Publishing 101 ($16.95) by Regina Smith is a crash course in self-protection regarding the confusing world of music distribution and publishing. The book begins with a history of copyright law and explains how Congress creates laws. Other chapters address technology's impact on copyright and fair-use laws, the World Wide Web, Webcasting, MP3s, and a brief chapter about sampling, followed by a discussion of copyright infringement.

Publisher-songwriter relationships and copyright protection are also covered. It concludes with lists of music-publishing organizations, libraries, and music conferences. No Walls Production and Publishing; tel. (888) 566-7915; Web www.musicpublishing.net.

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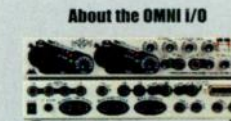

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KORG D1600

Korg's D1600 ($2,000) is the successor to the D16 Portable Digital Studio. The recorder not only retains the touch-screen interface but also adds faders for each channel, bays for swappable hard disks and a CD-RW drive, phantom power, and other new features.

The D1600 has 16 tracks of uncompressed 16-bit, 44.1 kHz audio, or 8 tracks in 24-bit, 44.1 kHz mode. In 16-bit mode, you can record eight tracks simultaneously; 24-bit mode lets you record four tracks at once. Each track has eight virtual tracks. Editing is nondestructive, with a selectable number of undo levels as high as 99.

As with its predecessor, the D1600 provides three bands of EQ per channel and built-in multi-effects algorithms. Configurations include eight insert effects for use with mono or stereo inputs and outputs, two master effects types with sends from each channel, and a stereo effect for the master out. Effects are programmable, and the unit has 192 user locations for storage. You can modulate insert effects with an expression pedal or a MIDI controller. One hundred scene memories can be recalled with Program Change messages, and automated mixing is available through MIDI Control Changes.

The D1600 supports MIDI Clock and MTC for synchronization. The unit sends and receives MIDI Machine Control so D1600s can be connected and synched.

The D1600's storage-bay system allows you to install IDE hard disks. You get a CD-R/RW bay for a Korg CD-RW-2 ($299) or a general-purpose ATAPI CD-R/RW drive. That lets you back up data, burn CDs, and import audio from CDs. The D1600's SCSI port can connect more storage units.

The D1600 has a 24-channel, 8-bus digital mixer with 16, 60 mm faders. Inputs include eight balanced ¼-inch jacks and one unbalanced ¼-inch jack for electric guitar or bass. Four XLR inputs provide +48V of phantom power. The unit also has two ¼-inch unbalanced jacks for both Master and Monitor outputs, and a pair of ¼-inch, unbalanced aux outs. S/PDIF digital I/O, with a built-in sampling-rate converter that handles 48 to 32 kHz sources, is also included. Other connectors include a ¼-inch footswitch jack, a ¼-inch expression pedal jack, and a ¼-inch headphone output. Korg USA, Inc.; tel. (516) 333-9100; Web www.korg.com.

YAMAHA MOTIF

Yamaha's Motif Music Production Synthesizer combines a high-quality sampling synthesizer and a digital-audio sequencer. Motif comes in 61-, 76-, and 88-key versions ($2,250, $2750, and $3,250, respectively) that offer 62-note polyphony, Velocity sensitivity, and Aftertouch.

The factory soundset includes stereo samples of acoustic instruments, dance-oriented sounds, ambient textures, and waveforms garnered from Yamaha's S80, 9000 Pro, and EX synths. If that's not enough, you can expand the unit with as many as three Yamaha PLG synthesis plug-in boards and 64 MB of sample RAM. Motif supports 16-bit stereo sampling at 44.1 kHz and accepts AIFF, WAV, Akai, and Yamaha A-series sample file formats. Sampling rates of 22.05, 11.025 and 5.5125 kHz are also supported.

You can digitally resample internal synth voices, performances, or sequences. Sampled phrases can be sliced into smaller chunks: Motif's Slice with Sequence feature analyzes the source material's amplitude and time characteristics to derive new audio segments and MIDI data. A phrase's feel, tempo, and dynamics can change without affecting the pitch of the performance. Sample with Note automatically maps samples to the keyboard.

Motif's 16-track sequencer accommodates MIDI and audio tracks, with room for 200,000 events. The sequencer supports pattern and linear recording, and song sections can be arranged, muted, and mixed in real time. Four knobs and four sliders handle a variety of assignable functions, including mixing and synth parameter control. The controllers' onboard templates offer a remote control surface for a host of popular software sequencers.

Motif has four ¼-inch unbalanced analog outputs; a USB MIDI interface; and MIDI In, Out, and Thru jacks. A SmartMedia slot and a SCSI port are included for data storage. The instrument has an expansion slot for the AIEB card ($249.95), which has AES/EBU I/O and six ¼-inch unbalanced outputs, or Yamaha's mLAN 8E card ($699.95). You also get ¼-inch unbalanced audio inputs for sampling, two expression pedal jacks, a breath-controller input, and sustain and footswitch inputs. Yamaha Corporation of America; tel. (714) 522-9011; e-mail info@yamaha.com; Web www.yamaha.com or www.yamahasynth.com.

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Both Motif and the RS7000 Feature an Integrated Sampling Sequencer (ISS)™ to Seamlessly Integrate MIDI and Audio Sampling

At Yamaha, we believe that making music shouldn't be work. So we designed the RS7000 and Motif Series keyboards with two goals in mind–inspiration and creation.

First, they were designed to inspire. Their sounds were hand-tailored by top sound design professionals from around the world. So whether it's an exacting replica of an acoustic piano for a Hollywood film score or a cutting edge sound from the clubs of London, these sounds were made by people with a passion for sonic perfection.

Next, we know that moments of inspiration need to be captured immediately. So all the tools for a complete music production are right at your fingertips. The features of the Integrated Sampling Sequencer, the hands-on user interface, and the built-in rhythm patterns, arpeggios and mixing templates were all focused on one goal–letting you create your music the way you want with the fewest technical obstacles.

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- –AIEB2 and Mlan Slots to Expand Digital and Analog I/O

So if you want to be on the leading edge of music production, check out these websites for more info and then check out Motif and RS7000 at a Yamaha dealer near you in June. Cause no matter how much money you make from making music, it shouldn't be work.

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YAMAHA DMI DIGITAL MUSICAL INSTRUMENTS

www.yamahasynth.com
www.motifator.com

* When converted to 16 bit linear format

WRH

SOUND ADVICE ▲▲▲▲

▲ NEMESYS

Electric slap-bass samples tend to be static due to the limitations of available sample RAM and the snapshot nature of sampling. Nemesys's *Scarbee J-Slap Bass* ($149; sample CD) takes advantage of the *Gigasampler* and *GigaStudio* hard-disk streaming engine to capture the nuances missing from most sampled slap basses.

Scarbee J-Slap Bass has 800 MB of sampled articulations including grace notes, hammer-ons, pull-offs, staccato releases, and slides, programmed in combination with muted, thumbed, and plucked notes. There are more than 900 sampled articulations devoted to three pickup settings: bridge, neck, and combined bridge/neck. Nemesys Music Technology, Inc.; tel. (512) 219-9181; Web www.nemesysmusic.com.

ROLAND

Two new sound libraries from Roland give the VP-9000 Vari-Phrase a selection of brass and reed phrases tailored for blues and swing. *VP-Z-05 Solo Instruments,* vol. 1 ($125) offers 208 tenor, alto, and soprano saxophone samples, including long tones, special effects, and phrases played over a variety of chord progressions. *VP-Z-06 Solo Instruments,* vol. 2 ($125) provides 234 trumpet, trombone, and flügelhorn phrases, including muted variations. You receive short licks and falls in addition to long tones, special effects, and articulations.

The sound libraries are shipped on Zip 100 disks and encoded with Roland's proprietary Wave Training. Roland Corporation U.S.; tel. (323) 890-3700; Web www.rolandus.com.

▼ POWER FX

Power FX's *The Electro Jazz Retro Funk Thing* recalls the halcyon days of '70s-era jazz-rock and fusion ($99.95, audio CD). The CD features a large number of vintage instruments, including Wurlitzer and Fender Rhodes piano samples. You also get drum and effects loops, construction kits, sampled chords, clavinet loops, synthesizer string pads, and drum hits.

The collection's sounds are patterned after the music of jazz-rock pioneers such as Herbie Hancock and the Headhunters. East West (distributor); tel. (800) 833-8339 or (212) 541-7221; e-mail sales@eastwestsounds.com; Web www.soundsonline.com.

AMG

Keith LeBlanc may be best known for his tenure as drummer with Tackhead and the Sugar Hill Gang. With AMG's release of *Out There* ($99.95, audio CD), LeBlanc steps out of the customary old-school hip-hop groove into African, Asian, and other ethnic rhythms as well as exotic experiments using a host of natural and processed sounds.

LeBlanc fuses his prodigious drum technique with science fiction–like sound effects, whales, cavern ambiences, and babbling brooks, using rock, ethnic, and orchestral percussion. The CD is also available as a package ($199) that includes two additional discs by LeBlanc: *Hip Hop Hard Phat* and *Old Skool Beats: Class of 2001.* East West (distributor); tel. (800) 833-8339 or (212) 541-7221; e-mail matt@amguk.demon.co.uk; Web www.amguk.co.uk.

▼ KEYFAX

On *Drums Around the World,* real drummers faithfully re-create global drum styles, which are captured as Standard MIDI Files (SMFs). Each file offers ten two-bar loops with performances mapped to General MIDI (GM) drum kits. However, savvy users can remap the performances to their drum sounds.

Drums Around the World includes performances in Brazilian styles such as bossa nova and olodum, Jamaican reggae, Cuban salsa (played by Latin-percussion whiz Michael Spiro), Japanese taiko drumming, and Senegalese drumming from the Malinke. Keyfax Software/Hardware; tel. (800) 752-2780 or (831) 460-0172; e-mail us@keyfax.com; Web www.keyfax.com. ●

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WEB PAGE

By Peter Drescher

WEB SITE OF THE MONTH

The **Vintage Synth Explorer** (www.vintagesynth.com) contains detailed information on more than 450 classic and modern synthesizers. Created in 1996 by Matt Friedman, the site is well designed, fun to look at, easy to navigate, and a valuable resource for any musician working with electronic instruments.

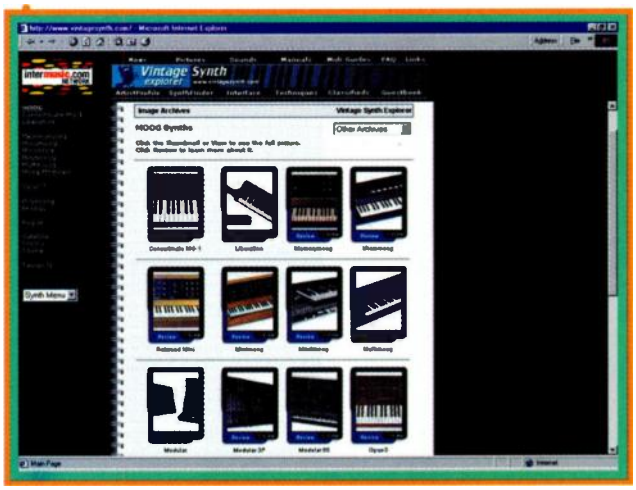
Synth connoisseurs will find bulletin boards, classified ads, lots of links, and plenty of eye candy. Interested in acquiring a Moog synthesizer? The Vintage Synth Explorer has pictures and descriptions of 20 models, including the venerable Minimoog and its predecessor, the Sonic 6. Besides technical information such as oscillator numbers, filter types, and manufacture dates, each synth description also includes links to feature-related instruments. Each instrument is given a rating of one to five stars based on visitor feedback, and many synth pages include RealAudio clips of the instrument's signature sounds.

Supplying minutiae on old synths is not the site's only purpose, however. Say you want to buy an instrument for a project in a specific genre. Use the SynthFinder feature to narrow your choices by stepping through a series of pages that guide you to the synth category that best suits your needs. Do you have an old analog synthesizer but aren't sure what those knobs are for? The Vintage Synth Explorer contains a large number of manuals in online and PDF formats.

But wait, there's more! The Vintage Synth Explorer features tutorials on interfacing your synth into your MIDI studio, FAQs about gear repair, links to a huge number of music- and synth-related sites, profiles on musicians such as Moby and Fatboy Slim, a community bulletin board, and a tips and techniques area. Be prepared to stay awhile when you visit this site.

DOTDOTDOT.COM

JazzRadio.org (www.jazzradio.org) is the official Web site of "Jazz from Lincoln Center," a series of concerts and educational programs from New York City's premiere nonprofit jazz-performance organization, Jazz at Lincoln Center. Hosted by Ed Bradley of CBS News, the programs feature commentary, interviews, classic jazz, and newly commissioned works performed by the Lincoln Center Jazz Orchestra under the direction of Wynton Marsalis. Currently, 35 one-hour programs are available on the Web as RealAudio streams. With titles such as "Gillespiana," "Ellington's Strayhorn," "Con Alma: the Afro-Cuban Big Bands," and "Speak No Evil: the Music of Wayne Shorter," JazzRadio.org really swings. . . . **HitSquad** (www.hitsquad.com) is an all-purpose musician's Web site out of Albion, Queensland, Australia. Among its many features is the Shareware Music Machine, a list of more than 3,300 music and audio applications for almost every computer platform (see "Web Page" in the May 2000 issue). The News section contains stories of interest to musicians, including software release



and update notices, articles about the latest Napster and MP3 controversies, and colorful rants by various musicians. References, tutorials, discussion forums, music books, sheet music, and links to related Web sites make HitSquad a valuable resource for the hobbyist and seasoned pro alike. . . . Another place to get your hands on music and audio software is **Sonic Spot** (www.sonicspot.com). You will find not only beta, demo, and shareware versions of various MIDI sequencers, digital-audio editors, and effects processors but also links to MP3 rippers, audio librarians, V.A.S.T. and DirectX plug-ins, patch editors, DJ tools, CD labelers, and other utilities. Most links include a feature list and screen shot. The search function lets you find what you're looking for quickly. Digital-audio and MIDI tutorials and a lively discussion forum are also available.

WEBCAST

National Public Radio (NPR) maintains a well-deserved reputation for exposing the American public to a wide range of music styles, genres, performers, and philosophies. As part of its millennial programming, NPR presented a series of segments highlighting the 100 most important American music pieces from the 20th century. Broadcast throughout the year 2000 on *All Things Considered*, *Morning Edition*, and NPR's weekend news magazines, the NPR 100 is now available on the Web in RealAudio format at www.npr.org/programs/specials/vote/list100.html.

So just how do you decide which are the most important musical pieces of the last century? NPR did it through a process of elimination. First, it posted a master list on its Web site of more than 300 works in a wide variety of styles—from Aaron Copland's *Appalachian Spring* and Bob Dylan's *Like a Rolling Stone* to Dizzy Gillespie's *Night in Tunisia* and the *Wizard of Oz* film score. Then NPR invited its listeners to log on and vote for their favorites. The list was further narrowed by a panel consisting of Wynton Marsalis, Kathy Mattea, Michael Feinstein, Isaac Hayes, and Don Dixon.

The NPR 100 is a testament to American diversity and musical genius. Each song in the list is examined in depth, not only for its compositional and lyrical content but also for its social significance. For instance, the segment on Samuel Barber's *Adagio for Strings* presents interviews with film directors Oliver Stone and David Lynch about the use of the piece in their movies; it is also noted that the music came to the forefront of the nation's consciousness when it was played at President John F. Kennedy's funeral. Featuring the sonorous voices of Noah Adams and Linda Wertheimer, the NPR segments are interesting, informative, and educational.

DOWNLOAD OF THE MONTH

The Tonos *TC8* ($29.95; www.tonos.com) is an online 8-track digital-audio recorder for the PC that is designed to work with the Tonos server, cleverly called the Collaboratory. The *TC8* lets musicians with Web access contribute tracks to online music projects. The application's simple design and easy-to-use interface make it a fun and interesting way to record over the Internet.

The minimum system requirements are not terribly rigorous: a Pentium II/266 MHz running Windows 95, 98, or 2000; a DirectX-compatible full-duplex sound card; 64 MB of RAM; and an Internet connection. To get started, register as a Tonos community member by selecting a user ID and password and then filling in your band name, the instrument you play, the styles you are interested in, and other details.

Next, download and install the *TC8* software and begin laying down tracks. You can apply effects to the audio (reverb, pan, gate, compressor, distortion, EQ, and chorus) using a series of simple controls and create a rough mix for others to listen to. When you're finished, upload the audio tracks and the mix information to the Project in the Collaboratory. Uploading can take a long time if you have a slow connection, even though the 16-bit, 44 kHz mono PCM audio files are compressed using the QDesign algorithm.

As a Project Manager, you can let the community at large have access to your project. That means that any registered Tonos users can download your audio and add their own tracks. Or you can add security codes to prevent users from collaborating without your permission. You can even post want ads on the Tonos site to find instrumentalists to work with. As a Community member, you can look through a list of available projects to see if anyone needs your unique talents.

Although Tonos was founded by some big music-industry names (including Grammy winners David Foster and Carole Bayer Sager and producer Kenneth "Babyface" Edmonds), the online manual's recommendation to plug microphones and guitars directly in to the back of your PC implies a

mostly consumer-oriented market for the *TC8*. On the other hand, an early success story revolves around 15-year-old Kristin Collins, whose discovery through the system led to her recording a song for last year's *How the Grinch Stole Christmas*. In addition, Tonos created the "Road to Fame" contest on VH-1 for discovering and showcasing new talent.

BAND ON THE WEB

Domenico Sciajno is an Italian double-bass player and composer working in an experimental new-music genre. His heavily sonified Web site (www.headroom.ws/slice1), designed by Barbara Sansone, relies on Flash, QuickTime, and MP3 files to demonstrate his sonic skills and repertoire. In his online biography, Sciajno notes that his interest in improvisation led him to explore the creative possibilities inherent in the interaction among acoustic instrument performance, indeterminacy, and live sound processing by electronic devices and computers.

The site's novel "scratching turntable" navigation system lets users explore Sciajno's music. Under Projects, he lists the A.I.R.S Quartet, a group of European improvisers (guitarists Christian Alati and Giuseppe Ielasi, percussionist Ruggero Radaele, and Sciajno) who have released a CD of their unpredictable musical landscapes called *I Am Surprised While It Is Actually Happening*. Formed in 1996, the A.I.R.S Quartet has played various new-music festivals and recorded the soundtrack for "The Mission," a radio piece by Heiner Muller that was broadcast on Italy's RAI station.

Also listed is an audio-visual piece called *Objectable*, which combines live performance with a real-time processing and sound synthesis system written in Cycling '74's interactive MIDI and audio programming environments *Max* and *MSP*. A QuickTime movie of the resulting chaotic images, screams, pops, whistles, beeps, percussive growls, electronic clanks, and metallic bumpings can be downloaded, as can a PDF file of the score.

You can also listen to MP3s of Sciajno's other compositions. Each piece contains a description of the music and the date and place it was recorded. For example, *Walking Piano* was performed in Turin in 1998 when Sciajno sampled a one-minute improvised acoustic bass solo into 36 zones on a keyboard and then processed the sounds live using a computer system running *Max*. The results are an astonishingly wide array of sounds, all derived from the original source material.

WEB APP

Open Sound Control (OSC) is a new communications protocol designed to let synthesizers, computers, and other audio devices talk to each other through modems, the Internet, and high-speed networks such as FireWire, USB, and Ethernet. Although OSC can be used to transfer MIDI data, it's meant to be an open, efficient cross-platform solution to some of MIDI's inherent problems. For example, OSC's data transfer rate is about 300 times faster than MIDI's.

OSC data is transferred in packets (aka datagrams): chunks of information wrapped in a standardized manner for delivery over a network. Packets allow audio data to be processed statelessly (no assumptions are made about the device's current or previous state). OSC also provides for synchronicity, a Jungian term that implies the simultaneous occurrence of events, such as multiple MIDI notes transmitted in the same message or SMPTE-like locking of multimedia graphics to sound.

Each data packet consists of a symbolic address and message name, followed by binary data of any length. The addresses bear a striking resemblance to a URL's backslash/word format. That lets devices linked together by a Local Area Network (LAN) communicate easily, and it also facilitates the downloading of data for a specific device from an Internet site. The hierarchical structure's open-endedness also avoids the limitations of a system that relies on fixed-length data fields, such as MIDI and ZIPI.

Recent tests of the OSC protocol using SGI workstations and Macs have resulted in satisfying real-time performances. In some ways, OSC is like the Domain Name System that provides Internet surfers access to Web sites by typing in addresses instead of strings of numbers. OSC's protocol provides composers access to electronic devices and instrument patches by name instead of channel and controller numbers, making creative networking easier.

We welcome your feedback. E-mail us at emeditorial@intertec.com.

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TC Electronic's M-ONE and D-TWO are very powerful yet affordable new multi-effects units from one of the most highly respected names in signal processing. I highly recommend the M-ONE and D-TWO
Mike Collins - Electronic Musician (US)

M-ONE FEATURES

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TC introduces the truly musical RhythmTap feature: Not only tempo, but actual rhythm patterns can be tapped directly - or quantized according to a specific tempo and subdivision. The delays and rhythm patterns can be up to 10 seconds each. Control the exact number of repeats with Absolute Repeat Control.

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- Absolute Repeat Control
- Up to 10 seconds of Delay
- 50 Factory/100 User presets
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- 1/4" Jacks - Dual I/O

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MULTITAP RHYTHM DELAY

A studio effects processor dedicated solely to providing digital delay is a rare thing in this age of all-singing, all-dancing multi-effects units. In the D-TWO, TC have produced a very attractive delay unit that sounds exemplary and offers some unique features as well as all of the expected ones.
Paul White - Sound On Sound (UK)

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Oakland, CA - Leo's Pro Audio
Totowa, NJ - Guitar Center

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Orlando, FL - Sam Ash
Dallas, TX - Brook Mays
Southfield, MI - Guitar Center
San Francisco, CA - Computers & Music
Springfield, NJ - Guitar Center

June 6th
Fern Park, FL - YRS MIDI
Central Dallas, TX - Guitar Center
Toledo, OH - Peeler Music
San Jose, CA - Guitar Center

June 7th
Tampa, FL - MARS Music
Arlington, TX - MARS Music
Toledo, OH - MARS Music
Hollywood, CA - Sam Ash
Cherry Hill, NJ - MARS Music

June 11th
Clearwater, FL - Sam Ash
Austin, TX - Music Makers
Orange, CA - MARS Music
Franklin Mills, PA - Sam Ash

June 12th
Ft. Myers, FL - Brent's Music
Austin,TX - MARS Music
Irvine, CA - Jim's Music
Bryn Mawr, PA - Medley Music

June 13th
San Antonio, TX - MARS Music
Jeffersonville, IN - Far Out Music
Cerritos, CA - Sam Ash

June 14th
Ft. Lauderdale, FL - MAE
South Bend, IN - Woodwind & Brasswind
Sherman Oaks, CA - Guitar Center
Cherry Hill, NJ - Sam Ash

June 18th
Ft Lauderdale, FL - MARS Music
North Houston, TX - Guitar Center
Southbay, CA - MARS Music

June 19th
West Palm Beach, FL - MARS Music
Houston, TX - MARS Music
New York, NY - B&H Photo

June 20th
Miami Lakes, FL - Sam Ash
Baton Rouge, LA - Live Music
San Diego, CA - Pro Sound & Music

June 21st
Kendall, FL - Guitar Center
New Orleans, LA - Guitar Center
San Diego, CA - Guitar Center

June 22nd
Cerritos, CA - MARS Music

June 25th
St. Louis, MO - MARS Music
Arlington Heights, IL - Guitar Center

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Burbank, IL - Sam Ash

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Steinberg
NORTH AMERICA

By Julian Colbeck

Korg M1

Produced: 1988–94
Made in: Japan
Korg design team led by Jun-ichi Ikeuchi
Number produced: 250,000
Synthesis system: AI synthesis
Price new: $2,166

Today's prices:		
	Like new	*$650*
	Like, it's okay for its age	*$475*
	Like hell	*$350*

When the Korg M1 was previewed at the 1988 winter NAMM show, no one predicted that it would become the world's best-selling synthesizer workstation. Subdued and unassuming with a rounded casing, small display (by today's standards), and paucity of front-panel hardware, the M1 offers less than heart-stopping visuals. Fortunately, its layout is simple, and most operations are easy to understand and execute. The onboard sounds are catchy, direct, and versatile. (If you're a product designer, reread the two previous sentences. In fact, cut them out and tape them to your CAD system. They reveal the blueprint for a best-selling synth, then and now.)

At the heart of the synth, pulse-code modulation (PCM)–sampled and synthesized waveforms can be shaped using conventional analog-style editing techniques. As many as eight Programs on the same or different MIDI channels can be linked into a Combination. Throw in a sequencer and a decent pair of digital signal processing (DSP) chips for good onboard effects, and there you have it: the Korg M1.

In 1988 the M1's factory programs—acoustic guitars that actually sound like acoustic guitars; haunting oboes; melting strings; sonorous basses; and fierce, chunky pianos—took the world by storm. Even today M1 sounds are pleasantly direct yet full of character. You can always count on an M1 to get you out of trouble.

Programs use one or two oscillators generating sounds plucked from the M1's 4 MB pool of waveforms. Single-oscillator Programs are 16-note polyphonic, and double-oscillator Programs are 8-note. Sounds range from full multisampled pianos to bells and pan flutes; snippets of recorded audio such as Koto Trem, Pole, and Lore (a sample of a jack-in-the-box being wound up, courtesy of Steve Winwood's keyboard technician); and synth waveforms from the Korg DSS-1 and DSM-1.

Three independent four-stage envelope generators (EGs), each controllable with Velocity and Aftertouch, modulate the filter, amplifier, and pitch. The pitch envelope is great for adding interest to the beginning of each note. You can apply the amplifier envelope to cut off all but the beginning of a ROM waveform and then use the truncated waveform as the front end of a double-oscillator Program; the koto multisample's initial tip is a good example of that technique.

The Velocity-sensitive lowpass filter offers keyboard tracking but no resonance. Korg labored over the M1's filter; in the original design, the filter's EG had no Intensity parameter. By the time it was redesigned, it featured positive and negative Intensity.

The M1 was one of the first synths to offer a serious collection of premapped drum kits. For its vintage, the drum samples are first-class. They can even be remapped, which is handy because the M1's kits don't adhere to what has become standard General MIDI (GM) mapping. You can also apply Pitch Bend to drum sounds.

Perhaps the M1's most confusing aspect is its flexible effects routing, because the effects and the overall output routing are inextricably bound. You have a huge

COURTESY KORG USA

The Korg M1 is the best-selling synth workstation in history. Its piano sound spawned myriad dance tracks in the mid-'80s. A subculture of sounds and support was based on the M1, as was a generation of sample-based clones from Korg and almost every other electronic musical-instrument company. It was the last big instrument without knobs.

Get In. Get Out. Take Control. Introducing the US-428.

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variety of choices, including routing the effects in series or parallel and sending each Program to a separate output.

A pair of digital multi-effects processors provide reverb, delay, overdrive, EQ, chorus, rotary speaker, and other effects. Effects can be applied to specific Programs within a Combination. The M1's effects are remarkable for their high quality and real-time controllability; they're clean, powerful, and completely editable. You can assign a footswitch to change an organ Program's rotary-speaker effect from slow to fast in real time.

You can adjust many of the M1's basic parameters—such as filter cutoff, envelope release, and effects level—by choosing the parameter onscreen and moving the Edit slider or a footpedal, also in real time. Eight primary parameters are on the main Program page; you gain access to them by pressing one of the round function buttons beneath the display.

It seems that only die-hards still use the M1's sequencer. It offers eight tracks, full quantizing and editing, and the option to store phrase patterns that can be inserted into a sequence. But when compared with a modern workstation-based sequencer, the M1's number-based user interface is hardly inviting.

The range and quality of its building blocks give the M1 character, which, in its time, resulted in substantial success for Korg. That success might be surprising when you consider that the filter has neither resonance nor modes other than lowpass, the low-frequency oscillator is basic, and no sync or cross-modulation exists between the oscillators. Fortunately, you can find PCM expansion cards with more complex types of synth sounds.

When the M1 was au courant, Korg and an army of third-party developers gave it royal support. Instructional materials supplemented new collections of PCM samples and Programs, such as Korg's MSC-1S-16S cards, which contained raw PCM data and Program/Combi data. Frontal Lobe produced the PCM Channel SysEx kit, which let users load their samples into an M1. InVision introduced the M1 Plus One expansion board, which had 4 MB of sparkling new samples such as organs, guitars, and flutes; those samples are preinstalled on the M1 Plus One. Some bloke named Julian Colbeck even made a fine video called *Getting the Most out of the Korg M1*.

The M1R is an otherwise identical rack-mount version. It was upgraded to the M1REX (commonly called Mirex), and its ROM waveform capacity doubled, with 275 multisampled sounds instead of the M1 keyboard's 144. Korg implemented an overflow mode, providing twice the polyphony by linking two instruments. The M1 can be upgraded to Mirex status, giving it the same waveform ROM as Korg's T-series.

In 2001 Korg's support for the M1 is limited to repairs and replacement of the original patch data. Unfortunately, the M1's annoying battery-changing routine wipes your Programs and Combis from its memory. If you externally back up your internal data through SysEx, though, you shouldn't have a problem. Luckily, the M1 is a reliable beast, with only the occasional sticky button to beware of.

If you're ever stuck, the M1's Internet presence is massive, with endless newsgroups and Web sites devoted to it. Many of the sites are tiresomely fan based, but a few offer reprints of manuals, helpful tips, and downloadable Programs. I particularly like Terry Little's Web site (www.geocities.com/tlittle72), which is succinct, relevant, and helpful.

Although other Korg instruments have superseded the M1, it still holds its own on a dance track. The reggae fraternity would, no doubt even today, be lost without the M1's skanking-friendly pianos and bubbling organ sounds. An M1 revival seems unlikely, though, because the instrument never went away.

Julian Colbeck *has toured everywhere from Tokyo to São Paulo with artists as varied as Yes, Steve Hackett, John Miles, and Charlie.* *Thanks to Jerry Kovarsky, Leslie Buttonow, and Jack Hotop of Korg USA.*

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TECH PAGE

By Scott Wilkinson

Semiprecious Semiconductors

When I'm not playing or writing about electronic music, I pursue several hobbies, one of which is mineralogy. Among my favorite types of minerals are those that exhibit interesting optical properties, such as fire opals, which glint intense red, green, or blue depending on the angle at which light hits them.

Fire opals may hold the key to photonic communications.

That phenomenon, commonly called opalescence but known technically as *Bragg reflection,* offers promising potential in the field of *photonics.* The goal of photonics is to manipulate photons in much the same way that electronic components manipulate electrons. If the fledgling technology is successfully developed, it could increase digital communication and processing speeds by several orders of magnitude.

When light strikes a boundary between two transparent materials, part of it is reflected and part is transmitted through the boundary. If the transmitted part encounters another boundary after the first one, the same thing happens again. If the spacing between the boundaries is exactly half the wavelength of the incoming light, the part reflected from the second boundary constructively interferes with the part reflected from the first boundary, which intensifies the light that finally reaches your eyes (see **Fig. 1**). If there are many equally spaced boundaries, all the incident light at the critical wavelength is reflected; that is Bragg reflection.

Opals exhibit Bragg reflection because they consist of many tiny spheres of silicate glass. Those spheres are tightly packed, much like oranges in a crate, and they measure several hundred nanometers in diameter, which is on the same scale as visible-light wavelengths from 400 to 700 nm. As a result, when white light strikes an opal from certain directions, the gap between the spheres is exactly half the wavelength of certain components of the light, and the eye sees dazzling green, red, or blue reflected from the stone.

If a material could be made that exhibited Bragg reflection from all directions at a particular wavelength, no light at that wavelength could enter from the outside and any light at that wavelength originating within the material would be trapped by endless internal reflections. Then, by introducing imperfections in the crystal lattice, the flow of photons into and out of the material could be controlled, much like the way in which impurities in semiconductor material allow electron flow to be controlled.

Many scientists are trying to create such materials. For example, Willem Vos at the University of Amsterdam is experimenting with suspending tiny polystyrene spheres in a liquid and letting them naturally settle into a crystal. After drying the crystal, he fills the air between the spheres with a highly refractive material, such as gallium arsenide, and heats the crystal to evaporate the polystyrene, which leaves a latticework of gallium arsenide surrounding spheres of air—a sort of inverse opal.

The key to unlocking the technology's potential is placing a microscopic light source within an inverse opal. That would trap the light until it could be released in a controlled manner through imperfections in the lattice. It hasn't yet been accomplished, but according to one pioneer in the field, physicist Sajeev John, "We are tantalizingly close."

If photonic circuitry can be developed to keep data in the form of light pulses rather than convert it from photons to electrons and back again, as it is in current communication systems, the pace of data handling will increase dramatically. The Internet will operate at World Wide Warp speed, and all sorts of processing, including musical applications, will benefit greatly. So the next time you admire an opal, remember how it might inspire the future of communications and computing.

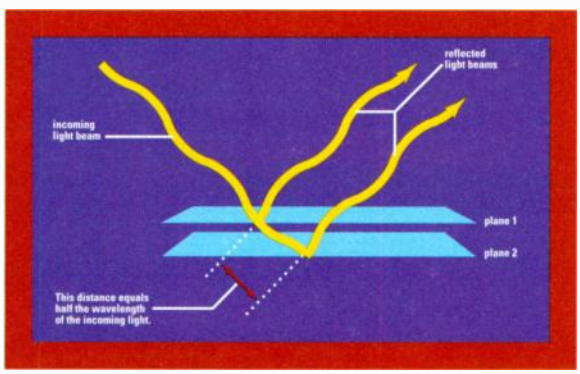

FIG. 1: When the distance that light travels from one transparent boundary to the next within a crystal is half its wavelength, the reflected portions constructively interfere, intensifying the light.

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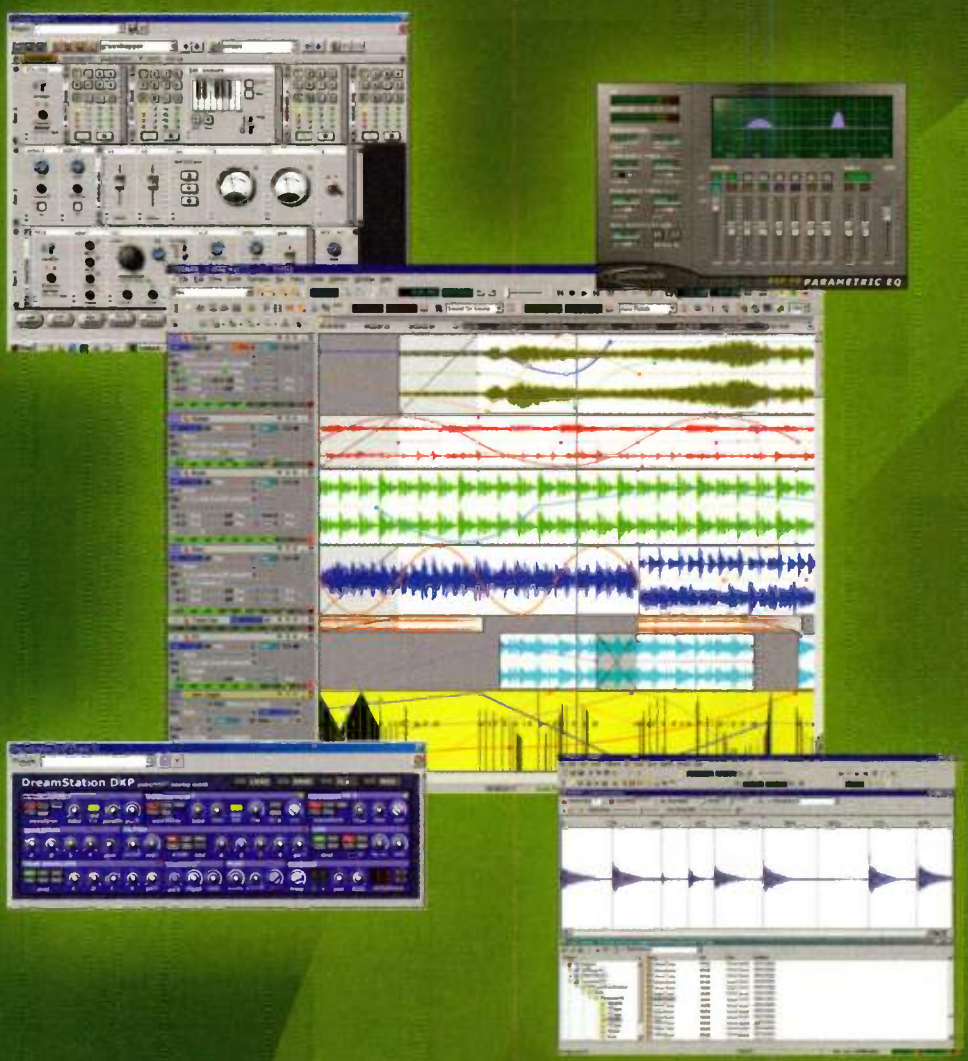

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- 4 programmable Chord Memory buttons for triggering chord voicings easily
- The ability to load new GEs as they are released

MAKING

5.1 MIXING ON A Budget

Mixing music in 5.1 surround is exciting, but the price of equipping your studio for the task might seem prohibitive. If you limit yourself to professional equipment, things can get expensive quickly. Does that mean that 5.1 surround mixing is the exclusive domain of the rich and famous, or is there a way to do it on a real-world budget? By adding some consumer gear to your existing project studio, you can mix 5.1 surround for a lot less than you might think. (If you aren't up to speed on the basic technology of surround sound, see "You're Surrounded" in the October 2000 issue; for mixing techniques, see "Mixing in the Round" in the May 2001 issue.)

Surround mixing needn't break your bank.

WIDE RECEIVER

To monitor 5.1-channel surround music in your studio, you need five full-range speakers and one subwoofer, amplifiers for the speakers and subwoofer, a volume control with bass management, and a Dolby Digital/DTS decoder so you can check that the final surround encoding worked.

Fortunately, a modern home-theater receiver (also called an A/V receiver) takes care of those major monitoring requirements (except the speakers and subwoofer) in one fell swoop. All home-theater receivers include Dolby Digital or DTS decoders (or both), which accept a digital bitstream from a suitably encoded disc in an appropriately equipped DVD player and then convert it to six audio channels. In the studio, that conversion ability is important only for checking a file that has been encoded; it is not used during the mixing process. However, make sure that your receiver has both decoding types, just to cover your bases.

BY MIKE SOKOL


ILLUSTRATION BY PIERRE-PAUL PARISEAU
SALE

More important for mix monitoring, all A/V receivers provide amplification for the five main channels, but few include an amplifier for the subwoofer, which means you'll need a powered sub. Many receivers also include preamp outputs, which let you use external amplifiers or powered speakers.

A large knob controls the volume of all six monitor channels simultaneously, and a bass-management filter directs frequencies below a certain point in the main channels to the subwoofer. Basic receivers have a fixed bass-management frequency (typically 80 Hz), but more expensive models let you select from a list of cutoff frequencies. To gain access to that and most other parameters, the receiver must be connected to a video monitor so you can see its onscreen-display (OSD) user interface.

For project studios, it's important to use a receiver with six discrete analog inputs. Those inputs are intended for the new DVD-Audio and SACD players, which have six analog outputs. In the studio, however, those inputs are perfectly suited for accepting the 6-channel output from the mixer or multitrack mixdown deck for monitoring purposes.

Basic A/V receivers that provide 60W of amplification per channel can be purchased for less than $300. You can also spend $1,000 to $1,500—or more—for a high-end receiver with time-delay correction (which compensates for speakers placed at different distances from the listener), calibration microphones (for automatically adjusting speaker levels), variable frequency bass-management filters, and other bells and whistles. Many companies make excellent receivers; check out the offerings from Denon, Kenwood, Marantz, Pioneer, Sony, Technics, and Yamaha. **Fig. 1** illustrates how to integrate a home-theater receiver into your surround-mixing system.

If your mixer outputs signals at +4 dBu, you need to know that consumer receivers are designed to accept –10 dBV signals at their inputs. To convert from +4 to –10 levels, you can build a simple resistive pad (see **Fig. 2**). If you're not a do-it-yourselfer, you can get a Line-Level Shifter from Ebtech (www.cymation.com) for about $200. That handy device changes the signal levels of eight outputs from –10 to +4 and vice versa. It also automatically converts from balanced to unbalanced (and back again) depending on whether you plug in a TRS- or TS-phone plug.

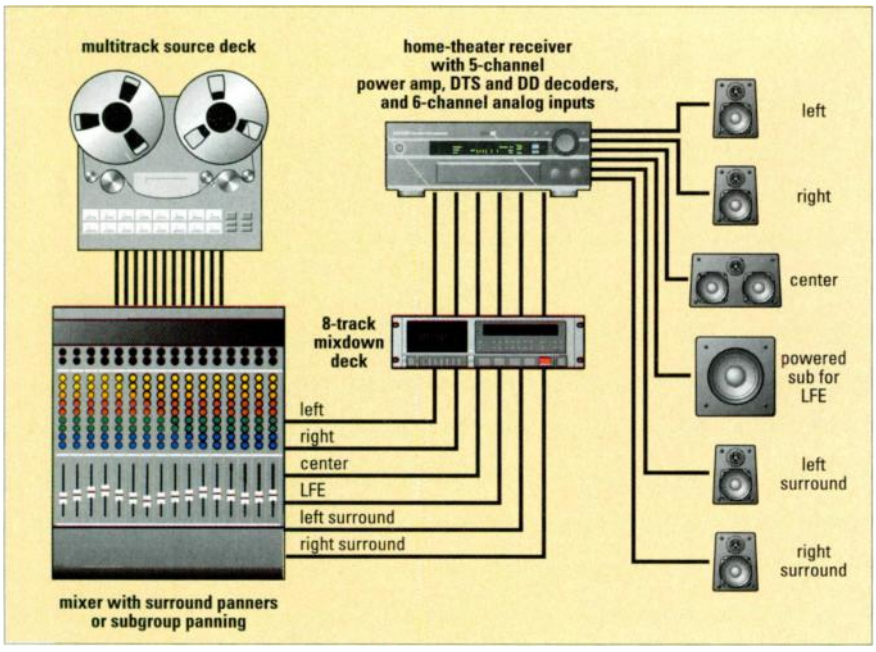

FIG. 1: A consumer home-theater receiver can serve well as a monitor amp and bass-management filter for a surround-music recording system.

SPEAK TO ME

I don't like to use consumer speakers for mixing in the studio because they tend to color the sound, which can disguise the real mix. I recommend using inexpensive passive studio monitors that connect to the speaker outputs of the home-theater receiver. You probably already have a pair of nice monitor speakers, so you might be tempted to simply add three less-expensive, alternate brand or model speakers for the center and surround channels. Don't succumb to that temptation; the best thing to do is match all five speakers.

For those on a budget, I recommend something such as the Alesis Monitor One MkII ($299 per pair). If you have a pair of the original Monitor Ones, buy three MkIIs for the front left, center, and right speakers, and move your original Monitor Ones to the left and right surround positions. Other possible choices for monitors include the Roland RSM-90 ($199 each) and the Tannoy Reveal ($399 per pair).

A powered subwoofer is a must, because few, if any, receivers provide amplification for the low-frequency effects (LFE) channel, and many powered subs are available for a reasonable price (see the sidebar, "Subs for Sale"). Buy a sub with as much power as you can afford; after all, it has to handle the LFE channel and the bass frequencies in all five main channels. Go for at least 100W, but the more the better.

CALIBRATION ON THE CHEAP

Once you have the receiver and speakers set up, it's time to calibrate the system. You can get a nice real-time analyzer (RTA) for $1,000 to $2,000, which makes calibration a piece of cake. But that's a lot of money when you can do the same basic job with a $50 Radio Shack sound-pressure level (SPL) meter, which thousands of recording studios worldwide have embraced as the cheap alternative to a high-priced RTA.

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Adjust the output level of each speaker while playing pink noise through it. Nearly all modern A/V receivers include a built-in pink-noise generator for setting the speaker levels properly. When that function is engaged, the pink noise automatically cycles from one speaker to the next at a predetermined rate, which means you may have to let it go around the room several times before you get the levels right.

Prior to beginning calibration, make sure that the correct amplifier channels are hooked up to the appropriate speakers. There's no better way to ruin the sound of your mix than by accidentally swapping the surround and front speakers, but I've seen it happen time and time again. So make sure everything's connected with the correct polarity from the receiver to the speakers.

Start calibrating by placing your SPL meter at the center position where your ears would normally be, pointing it directly at the speaker that is playing the pink noise (only one speaker at a time should be making noise). Modern receivers include parameters that let you adjust the level of each speaker in their OSD controls. If you're using five matched speakers equally spaced around the mixing position, the gain settings for the left, right, center, left surround, and right surround should all be the same, but the bottom line is that the SPL meter should read the same level from all five speakers (use C weighting). For movie sound, the generally accepted level is 85 dB SPL with the receiver's master volume set to 0 dB, but that's not critical for music.

The LFE level from the subwoofer needs to be 10 dB higher than the main speakers, but only within the two octaves it covers. When properly calibrated with an RTA, the LFE level will read 10 dB higher than the main speakers. However, a simple SPL meter averages the energy in those two octaves over the entire audible frequency range, which makes the LFE appear to be only 4 dB hotter when properly calibrated; set the receiver's LFE gain control so that the SPL meter reads 4 dB higher than the main speakers.

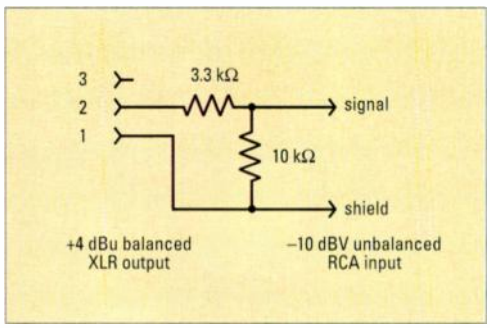

FIG. 2: If your mixer outputs signals at +4 dBu, build this simple resistive pad to convert the signals to –10 dBV, which is what the receiver expects.

Once your receiver and monitors are properly calibrated, verify that your mixing console is connected properly and calibrated. Start by injecting pink noise from a signal generator, test CD, or the console's built-in generator into a single channel strip. Patch the strip to all six surround output buses equally. (I typically set the console to individual-send mode rather than using the joystick or pan/fader steering). Then set each channel send to unity gain. The surround bus output faders should also be set to unity gain.

Next, go back to the input strip and set the trim so that the output level is –20 dB on the console's meters. That is called –20 decibels full scale (dBFS). Finally, make sure that the mixdown deck is also showing –20 dBFS on all six tracks; if not, trim the console's ADAT or TDIF outputs or the tape deck's input levels. (The LFE channel only gets an extra 10 dB of gain in the final monitoring part of the sound chain, not the recording path.)

SMALL EQUALS BIG SOUND

Modern A/V receivers include a parameter that lets you specify your main speakers' size—small or large. That setting determines whether the receiver's bass-management function is engaged. If you specify large speakers, the bass frequencies will be routed directly to the speakers without passing them to the subwoofer. But if you specify small speakers (even if your speakers are physically large), the bass below 80 Hz will be pulled from the main channels and redirected to the subwoofer, which is exactly what you want (unless you're using big speakers that reproduce down to 20 Hz at full level).

You want to enable bass management so that any bass problems can be heard and corrected. In a 5.1 surround–music mix, you don't need to put audio in the LFE channel. Any bass in the main channels will be routed to the subwoofer during playback anyway and at the proper levels. I've done music mixes with nothing in the LFE channel that have tremendous bass during playback. Nevertheless, what you put in the LFE channel is your business as long as you calibrate it properly in your studio.

Surround mixing needn't cost you an arm and a leg. With a consumer A/V receiver, five inexpensive monitors, a powered subwoofer, and a Radio Shack SPL meter, you can turn your stereo studio into a surround room for much less than the big shots typically spend. All that's left to add is your creative genius, and you're on your way toward the future of recorded music.

Subs for Sale

Manufacturer and Model	List Price
Definitive Technology PF15TL	$699
Hsu Research VTF-2	$499
Polk PSW350	$420
PSB Alpha SubSonic 1	$439
Signet SP100 II	$600
Snell QBx	$700
Tannoy PS110B	$499

Mike Sokol *is a human being, with 2.0 ears, learning how to mix in a 5.1 environment. For some reason, no one takes seriously his suggestion of using gene therapy to add 3.1 more ears to surround-mixing engineers.*

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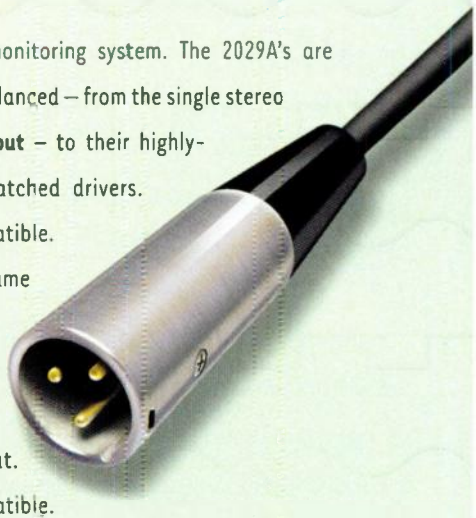

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WRH

GOOD REFERENCES

By Brian Knave

EM's guide to understanding and selecting close-field reference monitors.

Judging by the steady flow of letters and phone calls we get asking our advice about what gear to buy, a good number of readers are well acquainted with cognitive overload. That's the term psychologists use to describe the paralysis that can set in when we are confronted by too many options (or too much information). Freedom of choice is great, but clearly, too many options can bewilder. Case in point: the EM 2001 Personal Studio Buyer's Guide lists 40 companies presently offering reference monitors, with more than 200 models to choose from.

Bewildered? If so, you've come to the right place. This article will cover the various designs, components, and properties (including terminology) of reference monitors, as well as how they work—in short, all you need to know to make informed decisions when selecting close-field reference monitors for your personal studio. (Though many of the concepts discussed here apply equally well to monitors for surround arrays, those interested specifically in monitoring for 5.1 should also see "You're Surrounded" in the October 2000 EM.)

PRE ROLL

Speakers used in recording studios are called monitors and generally fall into two categories: main monitors and compact or *close-field reference* monitors. Mains, as they are called, are mostly found in the control rooms of large commercial studios, often flush-mounted in a "false" wall (called a *soffit*); close-field reference monitors are freestanding and usually sit atop the console bridge or on stands directly behind the console.


ILLUSTRATION BY WWW.LOLABEE.COM

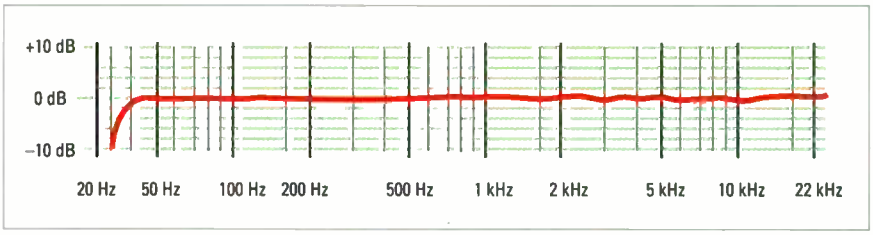

FIG. 2: Specs may not tell the whole story, but a frequency-response plot *is* helpful for assessing the spectral accuracy of a monitor. Shown is the published plot for a Mackie HR824 powered monitor. The written spec is 39 Hz–22.5 kHz (±1.5 dB)—impressively flat over a broad range.

Most personal studios don't have the space or funds for main monitors, so this article will focus on the compact reference monitor—a relatively recent studio tool. The first "compact" monitor to see widespread use in recording studios was the JBL 4311, a 3-way design introduced in the late 1960s. The 4311 was quite large, however (it had a 12-inch woofer, a 5-inch midrange speaker, and a 1.4-inch tweeter), and today would qualify more as a mid-field monitor.

As engineers increasingly realized the importance of hearing how their mixes sounded on car and television speakers, smaller reference monitors gained in popularity. One of the earliest favorites (around the mid-1970s) was the Auratone "cube," which had a single 5-inch speaker.

Car and home-stereo speakers kept improving, of course, so engineers were always on the lookout for better close-fields. One compact model that caught on big was the Yamaha NS-10M (see **Fig. 1**). A bookshelf-type speaker introduced in 1978 for home use, the NS-10M soon became a familiar sight in commercial studios, and it remains popular—or at least ubiquitous—to this day.

Another significant development was the introduction in 1977 of the MDM-4 near-field monitor, made by audio pioneer Ed Long's company, Calibration Standard Instruments. The MDM-4s were great monitors, but it was the then-revolutionary concept of near-field monitoring that secured a chapter in audio history for Long. (Long also originated the concept of time alignment for speakers and trademarked the term "Time Align"; more on this later.) Though no one could have predicted how prophetic the term *near-field monitor* would prove, Long clearly understood its significance and so had it trademarked. (That is why **EM** uses the term *close-field monitor* instead).

ENVIRONMENTAL ISSUES

Curiously, because close-field reference monitors have become increasingly accurate during the course of time, the original rationale for using them—to generate a good indication of how mixes will translate to low-cost car and home-stereo speakers—has waned. But there are also other good reasons close-field monitors have become all but indispensable in music production. For one, professional mix engineers are typically hired on a project-by-project basis, which means they may end up in a different studio from one day to the next. Close-field monitors, because they are portable enough to be carted from studio to studio, make for an ideal solution and guarantee, at the minimum, some level of sonic consistency, regardless of the room.

But don't the monitors sound different in different rooms? To a degree, they do. But another advantage of close-field monitors is that they can partially mitigate the effect of the room on what you hear. As their name makes clear, they are meant to be used in the "near field," typically about three feet from the engineer's ears. At that distance, assuming the monitors are well positioned and used correctly, the sound can pass to the ears largely unaffected by surface reflections (from the walls, ceiling, console, and so forth) and the various sonic ills they can wreak.

For the same reason, close-field monitoring is also a good solution for the personal studio, where sonic anomalies are the norm. As engineer, consultant, and all-around acoustics wizard Bob Hodas has so well demonstrated, however, it's foolhardy to think close-field monitors entirely spare you from the effects of room acoustics. "near-field monitors can be accurate," explains Hodas, "only if care is taken in the placement of the speakers and room issues are not ignored." (Find more information at www.bobhodas.com/publ.html.)

DIFFERENT WORLDS

A common misconception among those new to music production is that home-stereo speakers are adequate for monitoring. That is, in fact, not the

FIG. 1: The Yamaha NS-10M is arguably the most popular bookshelf-type monitor of all time. Despite its shortcomings, some pro engineers still rely on it regularly and can turn out great mixes using it alone.

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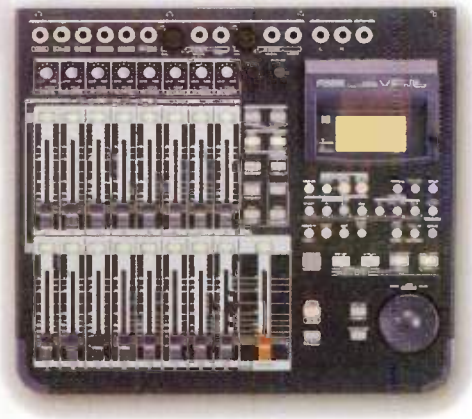

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case. The problem is one of purpose: whereas manufacturers design reference monitors to reproduce signals accurately, home-stereo speakers are specifically designed to make recordings sound "better." Typically, that perceived improvement is accomplished by boosting low and high frequencies. Although it may sound like an enhancement to the average listener, such "hype" is really a move away from accuracy.

Home-stereo speakers may also be engineered to de-emphasize midrange frequencies so as to mask problems in this critical range. That makes it difficult to hear what's going on in the midrange, which can tempt mixers to overcompensate with EQ. It can also lead to fatigue because the ear must strain to hear the mids.

Yet another reason home-stereo speakers are inappropriate for monitoring is that they are meant to be listened to in the far field, where much of the sound is reflected. But as we've seen, close-field monitors are designed to be used in the near field, in order to help minimize the effects of room acoustics. Of course, it's important not to sit too close to near fields. Rather, they should be positioned far enough back to allow the sound from the speakers to blend into an apparent point source and stereo soundstage. As you move in closer than three feet or so, the sound from each speaker becomes distinguishable separately, which is not what you want.

ELUSIVE BULL'S-EYE

Everyone can agree that reference monitors are meant to reproduce signals accurately. But what is accuracy? For our purposes, there are three objective tests that can be performed to help quantify accuracy in reference monitors. The tests measure *frequency response, transient* or *impulse response,* and lastly, *distortion.*

Frequency response is a measure of the changes in output level that occur as a monitor is fed a full spectrum of constant-level input frequencies. The output levels can be plotted as a line on a graph—called a *frequency response plot*—in relation to a nominal level represented as a median line typically marked 0 dB (see **Fig. 2**). The monitor is said to have a "flat" or *linear* frequency response when that line corresponds closely to the median line—that is, does not fluctuate much above or below from one frequency to the next.

When they are written out, frequency-response specifications first designate a frequency range, which is typically somewhere between 40 and 60 Hz on the low end and 18 to 22 kHz on the high end. To complete the specification, the frequency range is followed by a range specifier, which is a plus/minus figure indicating, in decibels, the range of output fluctuation. For example, the spec "50 Hz–20 kHz (±1 dB)" means that frequencies produced by the monitor between 50 Hz and 20 kHz will vary no more than 1 dB up or down (louder or quieter) from the input signal. (That spec would suggest a very flat monitor, by the way!) Note that the range specifier may also be expressed as two numbers, for example "+1/–2 dB," which is useful when the response varies more one direction than the other.

Primary frequency-response measurements are made on-axis, that is, with the test mic directly facing the monitor, often at a distance of one meter. Also helpful are off-axis frequency response plots (measured with the mic at a 30-degree angle to the monitor, for example), which give an indication of how accurate the response will be—or how much it might change—as you reach for controls or gear located outside of the "sweet spot." (The sweet spot is the ideal position to sit at in relation to the monitors; it is calculated by distance, angle, and listening.)

Transient or impulse response is a measure of the speaker's ability to reproduce the fast rise of a transient and the time it takes for the speaker to settle or stop moving after reproduction of the transient. Obviously, the first characteristic is critical to accurate reproduction of instrument dynamics and transients (such as the attack of a drum hit or a string pluck). The second is important because a speaker that is still in motion from a

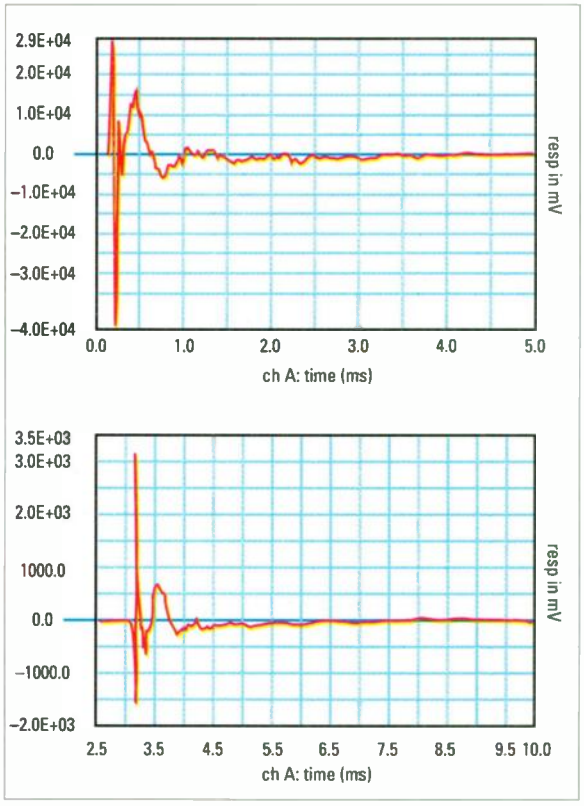

FIG. 3: In an impulse-response test, speaker movement is measured as the monitor is hit with an instantaneous signal. Shown are plots for monitors old and new: Yamaha NS-10M and JBL LSR28P, respectively. The first (narrow) spike is the tweeter, followed by the woofer spike (wide). Note how much the woofer response lags behind the tweeter in the NS-10M—and how jittery the decay response is—as compared to the LSR28P, which exhibits excellent impact and decay characteristics.

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previous waveform will mask the following waveform and thus muddle the sound (see **Fig. 3**).

Distortion refers to undesirable components of a signal, which is to say, anything added to the signal that was not there in the first place. For monitors it can be divided into two categories: harmonic distortion and intermodulation distortion (IM). Harmonic distortion is any distortion related in some way to the original input signal. It includes second- and third-harmonic distortion, total harmonic distortion (THD), and noise (which are the types most commonly measured; see **Fig. 4**), as well as higher harmonic distortions (fifth, seventh, ninth, and so on). Intermodulation distortion is a form of "self-noise" that is generated by the speaker system in response to being excited by a dynamic, multifrequency signal; typically, it is more audible and more annoying than harmonic distortion.

Frequency response, impulse response, and distortion levels should all be taken into account to get an idea of a monitor's accuracy. However, frequency response is often the only measure mentioned in product literature and reviews, and even it gets short shrift on occasion. (In many instances, I have seen frequency specs given with no range specifier—and of course, without it the specification is meaningless). Few manufacturers provide an impulse response graph (even assuming they have measured impulse response), and often the only distortion specification given is "THD + noise." In fact, the lack of established and agreed-upon standards for monitor (and for microphone) specifications—for both measuring them and reporting them—is a long-standing industry issue. Though it is true that specs don't tell the entire story, they are useful for corroborating what our ears tell us, and as such they can help educate us so that we can more exactingly listen.

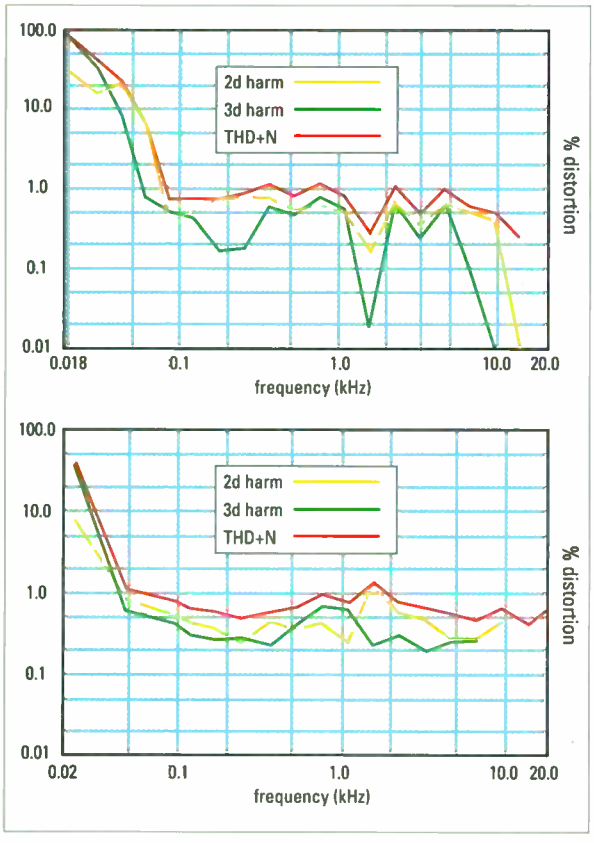


FIG. 4: Diagrammed above are two THD measurements for the two monitors discussed in Fig. 3, the Yamaha NS-0M and JBL LSR28P. Note the high bass distortion and moderate levels of second and third harmonics in the NS-10M.

MIRROR IMAGE

Now that we've established the *raison d'être* of the close-field monitor, let's take a look at its anatomy. We'll start with the internal components and work our way outward to the enclosure. Understanding how monitors are put together will help you know what to look for when deciding which best suit your needs.

Interestingly, the devices on either end of the recording signal chain—microphones and monitors—are very similar. Both are types of *transducers*, or devices that transform energy from one form into another. The difference is in the direction of energy flow: microphones convert sound waves into electrical signals and speakers convert electrical signals into sound waves. However, the components and operating principles of monitors and mics are essentially the same.

The speakers most commonly used in close-field monitors work in the same way as moving-coil dynamic microphones do, only in reverse. (Actually, there is a correlative speaker for other types of microphones as well, including ribbons and condensers. However, we will limit the discussion to the moving-coil type in this article.) In a moving-coil dynamic microphone, a thin, circular diaphragm is attached to a fine coil of wire positioned inside a gap in a permanent magnet. Sound waves move the diaphragm back and forth, causing the attached coil to move in its north/south magnetic field, thus generating a tiny electric current within the coil of wire.

In a loudspeaker, the coil of wire is known as the voice coil. As the electric current (audio signal) fluctuates in the wire, it generates an oscillating magnetic field that pushes and pulls against the magnet, causing the voice coil and attached diaphragm (in this case, the speaker cone; see **Fig. 5**) to vibrate. In turn, the vibrating speaker cone agitates nearby air molecules, creating the sound waves that reach our ears. (The ear, by the way, is also a transducer. It has a diaphragm—the timpanic membrane or eardrum—that converts acoustic sound waves into tiny electrochemical impulses which the brain then interprets as sound.)

DRIVING LESSONS

A loudspeaker's magnet, voice coil, and diaphragm form, collectively, an assembly called a *driver*. (The moving-coil

driver is the most common type, but there are other kinds as well.) Close-field monitors usually contain either two or three drivers, and thus are designated *2-way* or *3-way*, respectively. Standard 2-way monitors contain a woofer and tweeter; standard 3-ways contain a woofer, a tweeter, and a midrange driver. The woofer, of course, reproduces lower frequencies and the tweeter, the higher frequencies.

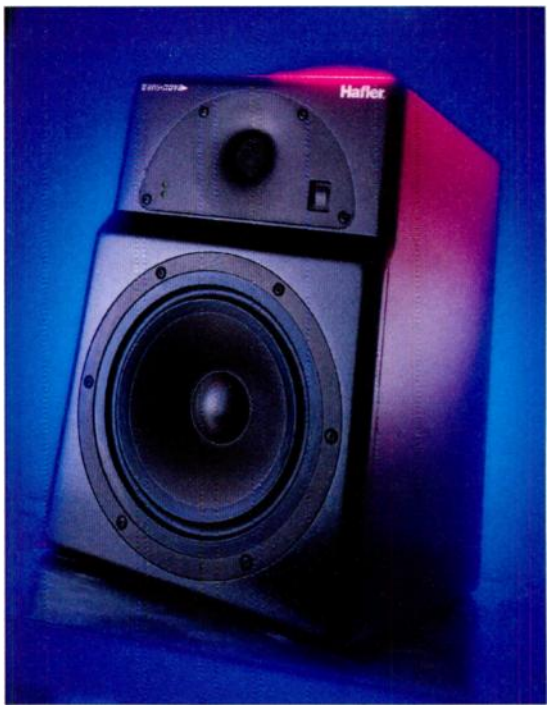
Cones and *domes* are the two most common types of diaphragms used in monitor drivers. Woofers and most midrange drivers employ cone diaphragms, typically made of treated paper, polypropylene, or more exotic materials such as Kevlar. (Note that the dome-shaped piece in the center of a woofer cone is a dust cap, not a dome.) Most moving-coil tweeters use a small dome, typically measuring one inch in diameter. One advantage of a small dome is that it exhibits fast transient response and a wide dispersion pattern, both of which are critical to the reproduction of upper frequencies. Domes are routinely made of treated paper too, but may also be made from a metal such as aluminum or titanium, or sometimes from stiffened silk, which some people believe sounds less harsh than metal.

When monitors employ separate drivers, as 2-way and 3-way monitors do, the design is termed *discrete*. In discrete designs, the drivers are usually mounted on the front face of the enclosure as close together as possible, which helps the sound blend into a coherent point source at the sweet spot. Depending on the monitors, the sound can change dramatically as you move away from the sweet spot.

IT'S ABOUT TIME

Some companies, for example Tannoy, employ an alternative driver design in some of their monitors in which the tweeter is mounted in the center of the woofer cone (see **Fig. 6**). Though more expensive, this *coaxial* design is naturally more time coherent than discrete designs because the drivers are positioned on the same axis (as well as closer together). Indeed, the coaxial driver arrangement is one of the design elements (among others) that manufacturers have used to meet Ed Long's Time Align specification, mentioned before.

Before we can understand how time alignment can improve a monitor's accuracy, we must first understand the timing problems inherent in conventional monitor designs. Discrete loudspeakers cause minute delays that spread sounds out in time, resulting in lost detail and a blurred or smeared sound. Specifically, sound from the woofer is delayed more than sound from the tweeter. This problem has two main sources, one structural, the other electronic. In a discrete monitor with a flat-face enclosure, the woofer voice coil is naturally set back further than the tweeter voice coil because of the extra depth of the cone in relation to the dome. The tweeter is therefore closer to your ears, causing the high frequencies to arrive slightly ahead of the lows.



With its round, beefy lows, smooth highs, and punchy mids, the Hafler TRM8 quickly garnered multitudinous industry raves—including an EM Editors' Choice award in 1999.

The problem is compounded by the *crossover*, an electronic circuit that splits the incoming signal into separate frequency bands and directs each band to the appropriate driver (more on crossovers momentarily). As it happens, crossovers also tend to delay low frequencies more than highs.

With his Time Align scheme, Long was the first to specify corrections for these problems, including physically lining up the drivers and adjusting driver and crossover delay parameters. When correctly implemented, Time Alignment ensures that the time relationships of the fundamentals and overtones of sounds are the same when they reach the listener as they were in the electrical signal at the input terminals of the monitor.

Over the years, some manufacturers have devised their own time-alignment schemes. You may recall, for example, the now-discontinued JBL 4200 series monitors, which employed protruding woofers designed to deliver low frequencies to the listener's ears simultaneous with highs from the tweeters.

WHEN I CROSS OVER

As mentioned, the crossover's job is to divide the incoming signal into separate bands and then send each band to the appropriate driver. In inexpensive monitors, this is typically accomplished using simple lowpass and highpass filters that split the signal coming from the power amp. This is called a *passive* crossover. In more sophisticated systems, an *active* crossover splits the line-level signal *before* it gets to the power amp. This requires each driver to have its own power amp, and is called biamping in 2-way monitor, triamping in a 3-way, and so on.

Typically, monitors that have active crossovers incorporate internal power amps. These are called *powered* monitors. The terms *active* and *powered*, though often used interchangeably, actually refer to different things: active

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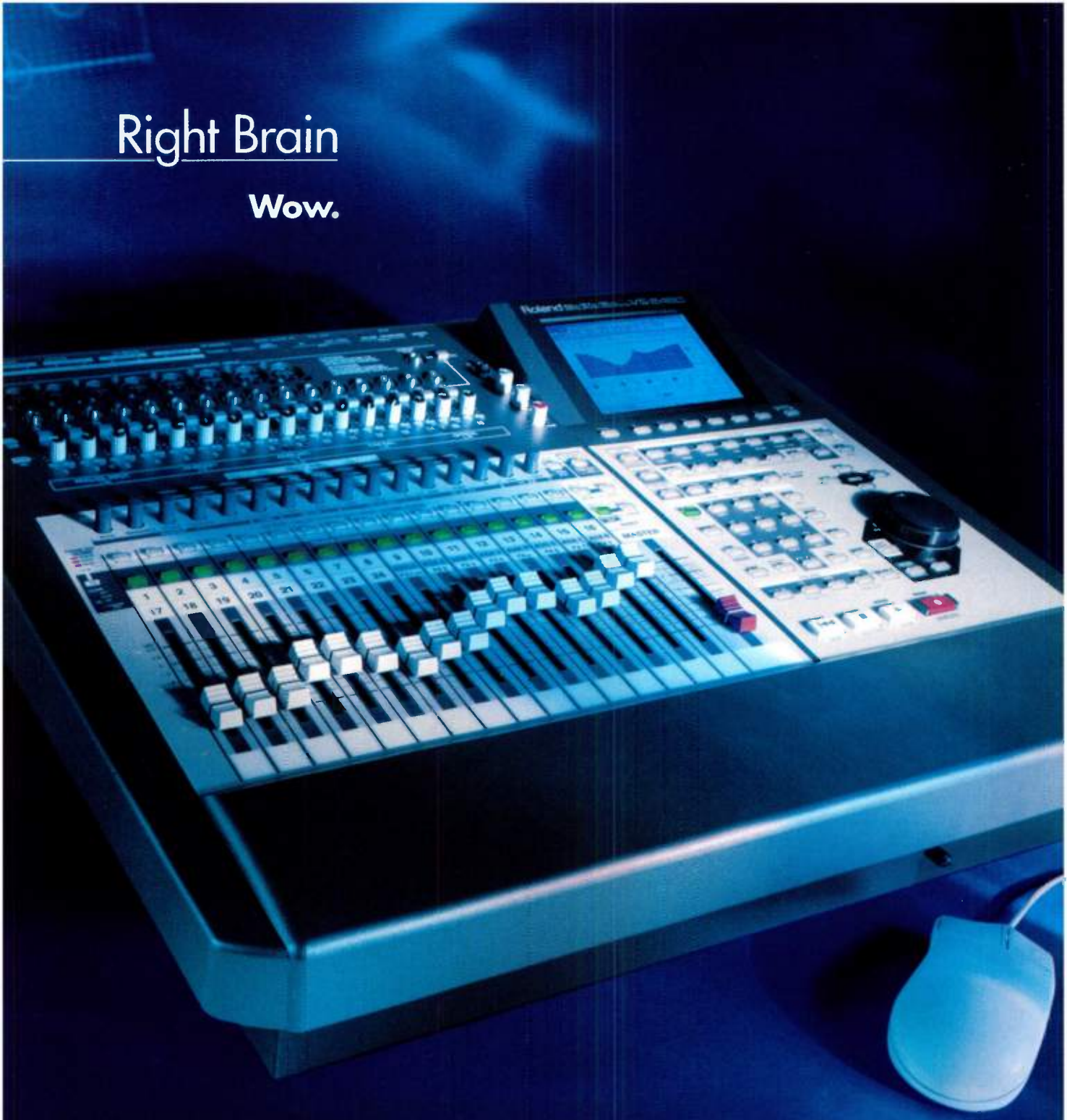

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refers to the crossover, and powered to the fact that the amplifiers are part of the package. In other words, although active monitors are almost always powered, not all powered monitors are active For example, Event Electronics at one time offered three versions of its popular 20/20 monitors: the straight 20/20 was unpowered and had a passive crossover; the 20/20p was powered but used a passive crossover; and the 20/20bas (biamplified system) was both powered and active.

In addition to giving a more exacting crossover performance, powered, active monitors offer other advantages over passive designs. Perhaps most importantly, because the amps and electronics are specifically designed to match the drivers and enclosure, powered monitors eliminate the guesswork and the potential pitfalls of matching an external amp to your monitors. (For a discussion of matching power amps to passive monitors, see the sidebar "A Good Match.") This means reduced risk of blowing the drivers and virtually no risk of overtaxing the amps. In addition, the internal wiring is much shorter, which cuts down on frequency loss, noise induction, and other gremlins attributable to long cable runs. The upshot is that a powered, active system provides a more reliable reference—no matter where you take the monitors, you can be sure the only variable is room acoustics.

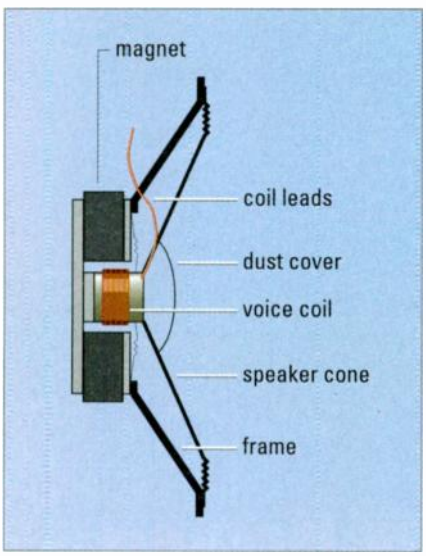

FIG. 5: A typical speaker driver includes a voice coil, permanent magnet, and diaphragm, which can be a cone or dome.

FIG. 6: Tannoy calls its range of coaxial monitors Dual Concentric. The System 800, above, employs a dual port on the front baffle.

BOX SET

The enclosure is a critical part of any reference monitor design. Compact monitors present a particular challenge to designers because diminutive enclosures do not support low frequencies well. For many small monitors, the lowest practical frequency is around 60 Hz. However, certain techniques allow manufacturers to extend the low-frequency response of their boxes.

A common solution is to vent or port the enclosure (see **Fig. 6**). The concept of porting is quite complex, involving not only one or two visible holes, but also other acoustic-design constructions inside the cabinet. In this design, often termed a *bass reflex* system, the port helps "tune" the enclosure to resonate at frequencies lower than the woofer's natural rolloff. That is, as the frequencies drop below the monitor's lowest practical note, the enclosure begins to resonate at yet lower frequencies, essentially providing a bass "boost." Although porting can extend the low-frequency response of the monitor well below a similarly sized but completely sealed enclosure (called an *infinite baffle* or *acoustic suspension* design), some people feel that the resulting bass extension is not a trustworthy reflection of what is really going on in the low end. (One noteworthy solution here is the incorporation of a subwoofer.)

Ports tend to be round, ovular, or slit-shaped, and usually are located on either the front or rear panel of compact monitors. Rear ports allow for a smaller front face, and therefore a more compact monitor, but they can also lead to sonic imbalances—the main one being excessive bass—in cases where the monitor is mounted too close to a wall or corner. Front ports help avoid this problem, but require a larger front face on the enclosure.

Another problem with front ports is that they can reduce the structural integrity of the front baffle (which is already weakened by at least two large holes, one each for the woofer and tweeter). Some ported monitors provide port plugs, which can be helpful for reducing low-frequency output in case you are forced to mount the monitor near a wall or corner. (A different solution for this problem is increasingly found in powered/active monitors—"contour" switches that let you adjust the monitor's low- and high-frequency output to compensate for acoustical imbalances in the listening space.)

Nowadays, most manufacturers build their enclosures from medium-density fiberboard (MDF), a material that offers better consistency and lower cost than wood. Grille cloths may or may not be provided with the monitors;


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but these are a cosmetic enhancement at best, and traditionally are removed for monitoring.

Because an enclosure's front baffle shapes the sound as it leaves the drivers, all aspects of the baffle must be taken into account by the designers. For this reason, designers often round off corners and sharp edges, and the face of the enclosure is kept as smooth and spare as possible in order to minimize interferences like diffraction (breaking up of sound waves). One critical acoustic-design feature on the front baffle is the *wave guide*—a shallow, contoured "cup" surrounding the tweeter. The structure and the shape of the wave guide both affect high-frequency dispersion, which in turn affects other sound qualities such as imaging (see **Fig. 7**).

PERFORMANCE ISSUES

Now that we've laid the groundwork, let's tally up what constitutes a superior monitor. Specifically, what do you hear in better monitors that you don't hear in lower-quality ones?

We already know one answer: accuracy. More than anything, the purpose and goal of a reference monitor is to transduce signals accurately. Monitoring is the last step in a long journey through the various processes required to get your music to its destination. Therefore reference monitors are your ultimate "feedback" system and the basis of all of the decisions you make about how to shape and process a mix.

As we've seen, the technical recipe for accuracy has three basic ingredients: accurate frequency response, accurate impulse response, and low distortion. Superior monitors boast a very flat frequency response, typically within ±3 dB of a nominal level. In addition, the frequency response should roll off smoothly at either end of the spectrum, as well as fall off evenly as you move away or off axis from the monitor.

Also critical is a monitor's impulse response. Ideally, this should be a direct analog to changes in air pressure in response to transient electrical signals; a superior monitor keeps all the "time domain" qualities of a signal intact, reproducing them in exactly the same time relation as they appear at the monitor's input terminals. In addition, in a superior monitor the frequencies issuing from discrete drivers are time aligned so as to compensate for the time misalignment inherent in discrete designs, as described earlier. That way, the highs, mids, and lows reach the listener's ear simultaneously.

Both impulse response and time alignment (among other things) figure prominently into two other critical sonic qualities of a reference monitor: *soundstage* and *imaging*. Soundstage refers to the imaginary stage that forms between two speakers (including width and depth), and imaging refers to how well the monitors can localize individual instruments on the soundstage. Obviously, a good soundstage and precise imaging are necessary for accurate positioning of instruments within the stereo field.

Distortion levels vary considerably from system to system. Whereas home-stereo speakers typically exhibit as much as 1 percent distortion above bass frequencies, some high-quality reference monitors may deliver as little as 0.1 percent. Though a low distortion spec is always desirable, some monitors with less-than-spectacular distortion specs still excel thanks to superiority by other measures. The human ear, however, is very sensitive to distortion, especially in the midrange (distortion is often a major contributor to ear fatigue).

Another helpful specification is speaker *sensitivity* or *efficiency*, which shows the monitor's output sound pressure level (in dB SPL) at a distance of 1 meter with an input signal of 1W. All things being equal (which they rarely are), speaker sensitivity has no determining effect on sound quality. However, if you are doing an A/B comparison of two or more sets of passive monitors and running them from the same power amp through a switching box, it is important to be aware of differing sensitivities. Our ears can readily perceive even slight differences in SPL, and our brains naturally perceive louder sources as sounding better. If you fail to compensate for any sensitivity differences—that is, to ensure that each monitor is playing back at the same level—you are more prone to reach incorrect assessments of monitors while comparing them.

FAITHFUL TRANSLATOR

Accuracy is important because, ostensibly at least, it guarantees that what we hear from our monitors is the "audio truth." Unfortunately, though, objective measures don't really guarantee accuracy. As helpful as specs may be, they are not really an indicator of how a monitor sounds; two similar

FIG. 7: One of the most impressive new monitors in years, the JBL LSR28P (powered, active) boasts an extremely flat frequency response over a broad range. Note the elliptical tweeter wave guide, which helps widen high-frequency dispersion.

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monitors with near-identical specs can sound very different, for example. Therefore, as in all things audio, careful listening must be the final measure. After all, monitoring is inherently subjective.

But even if monitoring weren't subjective and reliable standards for accuracy could be decided on and agreed upon, the problem of wide-ranging sonic differences among playback systems would still persist. More important than accuracy is knowing how your mixes will translate to other speakers in other environments. That's the real bottom line. And the only way to gain that certainty is from experience. As they say, practice makes perfect—and it's no different with reference monitors than with musical instruments. After all, a monitor *is* a musical instrument of sorts. Thus the need to spend many hours, many days, many months working with a set of monitors, "practicing" on them, listening to your results on countless playback systems, always fine tuning, adjusting, figuring out what the quirks are, where the bumps and holes are, and how every little thing translates, until you reach a level of familiarity that allows you to work undaunted, confident that the mix you dial in will bear a strong resemblance to what the end-user ultimately hears. Regardless of what monitors you use, until you are intimately familiar with them, mixing will remain something of a guessing game.

This point was brought home to me recently as I chatted with ace mix engineer Chris Lord-Alge. With multiple platinum credits to his name, Lord-Alge certainly qualifies as an "expert" on the subject of monitoring, at least in the sense that he knows what it takes to turn out mixes that sound great across the board, from boom box to high-end audiophile system. And just as surely, Lord-Alge has attained success enough to acquire and use any monitor he wants. So what monitors does he use? The latest, greatest, most expensive ones available? Not at all. Rather, Lord-Alge uses the same monitors he has mixed on for most of his career: a pair of Yamaha NS-10Ms. "The key thing with any monitors," explains Lord-Alge, "is that you get used to them. That's ultimately what makes them work for you. And 25 years on NS-10s hasn't led me wrong yet."

CAN OF WORMS

This brings us to a can of worms I'd just as soon not open—but open it we must if we're to inquire seriously into the nature of reference monitoring. Anyone who has searched for the "perfect" monitor has run smack into this dilemma, which is best summed up by these questions: Who, ultimately, are you mixing for? The snooty audiophile with speakers that cost more than most folks' cars? Or the masses who listen to music on cheap systems?

Lord-Alge's answer is enlightening: "Ninety-five percent of people listen to music in their car or on a cheap home stereo; 5 percent may have better systems; and maybe 1 percent have a $20,000 stereo. So if it doesn't sound good on something small, what's the point? You can mix in front of these huge, beautiful, pristine, $10,000 powered monitors all you want. But no one else *has* those monitors, so you're more likely to end up with a translation problem."

Similarly, I learned a few years ago that John Leventhal, who was one of my heroes at the time, did the bulk of his mixing on a pair of small Radio Shack speakers. (Leventhal, a New York City–based guitarist, songwriter, and engineer, made his mark by producing Shawn Colvin's acclaimed 1989 record, *Steady On*.) Leventhal owns both a pair of Yamaha NS-10Ms and a

A GOOD MATCH

Speakers are rated as to the amount of power they can handle from an amplifier. Like power amps, most speakers include a continuous and peak power rating. Make sure your amp and speakers have similar power ratings. You might think it's safer to use an underpowered amp than one that is overpowered. However, underpowered amps are more likely to be driven beyond their rated output, which results in clipping and generates distortion that could damage the drivers. It's much better to use an overpowered amp, as long as you exercise caution. Such an amp can easily supply enough continuous power with minimum distortion, and it has plenty of power in reserve (which is called headroom) for quick transient peaks.

Another important speaker spec is impedance, which is measured in ohms (Ω). Impedance is the electrical resistance the speaker presents to the current from the amp. The higher the speaker's impedance, the less current it draws from the amp. As the impedance drops, the amp must work harder to supply the current drawn by the speaker. In addition, the power output of an amp drops when it's connected to high-impedance speakers. For example, a particular power amp might deliver 100W to an 8Ω speaker or 200W to a 4Ω speaker.

Many speakers have an impedance of 8Ω, whereas others have an impedance of 6Ω, 4Ω, or even 2Ω. Most amplifiers can handle speakers with different impedances, but if the impedance drops too low, the amp could be damaged, so you should make sure that the impedance levels of your amp and speakers match.

—*Scott Wilkinson*

[Excerpted from *Anatomy of a Home Studio*, published by EMBooks, an imprint of artistpro.com, LLC.]

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pair of Radio Shack Optimus 7s. But he prefers the latter. "I love the Optimus 7s and use them more than the NS-10s. They're a little hyped around 4.5 kHz but they're more detailed than the Yamahas. I've discovered that if I get a mix sounding good on the Optimus 7s, it tends to sound good on everything else."

Perhaps now you see the dilemma I was talking about. This actually is a divisive subject, a point of contention among many engineers and music-production insiders. Some contend that accuracy is everything, and thus are ever on the lookout for the latest, the hippest, and—most importantly—the flattest monitor they can find. Others maintain that translation is the key, and thus they take a lowest-common-denominator approach, preferring to use monitors that better approximate what the general populace listens on.

Fortunately, there's an easy solution, and you don't have to choose sides. Instead, have it both ways. Set up, at the very least, a dual-monitoring system: one pair of quality reference monitors that you like the sound of and that you have determined to be reasonably accurate, and another pair of cheapies to represent the masses.

In my studio, I use a pair of Vergence A-20s (powered and active) for my main monitors, but I would never think of finalizing a mix without also checking how it sounds on my cheap Panasonic boom box, which is patched in and ready to go. The A-20s provide the flat response, high resolution, exacting transient response, and excellent imaging I need to make critical, often finicky mix adjustments, but typically it's the boom box that helps me see the forest for the trees and steers me clear of "macro" problems such as too much low midrange or not enough reverb. I also have a pair of Yorkville YSM-1s (an exceptional value, by the way) in my living room, and I regularly check my mixes there, as well as in my car. Trust me, multiple referencing is the way to go.

ROAD TEST

As you probably gathered by now, it's my opinion that familiarity with your monitors is more important than which particular ones you use. Not only has this been my experience, but I have heard the same time and again from other engineers. Still, that does not mean I recommend you buy just any monitors, on the grounds that you'll eventually get used to them. Rather, I recommend a thorough, methodical selection process, at least when choosing your primary reference monitors (the ones you expect to be accurate).

First, limit your choices by personal criteria. For example, you may be interested only in powered, active monitors, in which case you can rule out passive models. Or perhaps you have a budget constraint, in which case you can ignore anything above a certain price. Or you may know that you want, for example, a biamped design with woofers that are no smaller than six inches in diameter. (By the way, based on **EM** tests and reviews, we generally do not recommend multimedia speakers as primary monitors because the small woofers—usually five inches or less in diameter—don't typically provide sufficient low-frequency content for critical mixing.)

Once you've whittled the list down to five or ten models, it's time to deepen the research. Read reviews, check out online news groups (rec.audio.pro is a good one), and talk to pro-audio–savvy friends. Hopefully, this will help you pare down your list of monitors under consideration to a half-dozen or fewer models.

The next step is finding some way to listen to and, if possible, compare the contenders. A pro-audio store is your best bet for a place to set up a comparison test. Acquaint yourself with the staff and let them know what you want to do and hope to accomplish. Make sure the store has a decent listening environment, as well as a way to switch among pairs of monitors. Also, find out what type of playback source is available (CD player, DAT machine) so you'll know what format you should bring with you.

Before the day of the listening session, spend some time selecting appropriate reference music. One-minute excerpts are usually sufficient; if possible, burn

The acclaimed Genelec 1031AP helped popularize powered/active close-field reference monitors for the high-end commercial-studio market.

a reference CD for this purpose. An alternate idea would be to use an already prepared disc such as the "Mix Reference Disc," available from www.mixbooks.com. Choose tracks you are very familiar with, whether from commercial releases, projects you have mixed yourself, or both. Shoot for a broad sampling. For example, bring a classical orchestra piece with a huge dynamic range; a solo piano track; a section of spoken word; a traditional jazz recording; a heavy, full-band rock or pop tune; a folky acoustic guitar and harmony-vocal number; and, of course, be sure to include something representative of the style you primarily work in. You might also throw in a mix with a drum solo so you can hear clearly transients, high-harmonics from cymbals, lows from kick and toms, and the like.

As mentioned previously, it's imperative that each monitor play back at the same volume, so use an SPL meter to adjust levels to within 1 dB of each other. Begin listening in mono by panning everything to one or the other channel (as opposed to using a sum-to-mono switch, which puts the sound equally into each monitor); it's much easier to get an "objective" sense of a monitor's sound quality when listening to a single source rather than to a stereo field. When listening in stereo, make sure you and the monitors are positioned in the correct near-field relationship (see the sidebar "Now What?").

NOW WHAT?

Once you have selected your monitors, it's time to place them in your studio. An excellent starting point is to form an equilateral triangle with the two monitors and your head in the normal listening position (see **Fig. A**). The distance from each monitor to your head should be approximately three feet. You can increase the distance between the monitors, if necessary.

To allow more flexibility, try to mount the monitors on separate stands, not directly on the console or meter bridge of the mixer. Make sure the stands are solid and secure the monitors by using rubber feet or other nonslip measures. Small, lightweight speakers can actually crawl around when reproducing loud, low-frequency material if they are not secured in some way.

Angle the monitors inward toward your head and adjust their height so the tweeters are at the same level as your ears. Adjust the angle of the monitors to maximize the width of the sweet spot at the listening position.

Try to keep the monitors at least two to three feet from any wall or corner, particularly if they are rear-ported. (Alesis designers claim that anything more than six inches is okay.) If they are too close, the bass response will be emphasized. You might want to use port plugs to reduce the bass response if you must place the monitors near a wall or corner. Also, place the monitors far from any reflecting surface, thereby eliminating any external influence on the speaker's frequency response. (This is called placement in the free field.)

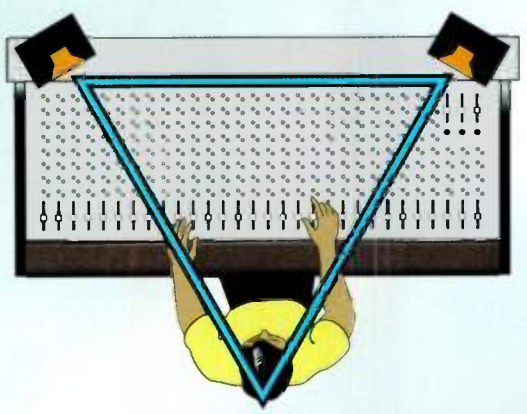

FIG. A: Close-field reference monitors are typically positioned to form an equilateral triangle with the engineer's head.

Another important technique is minimizing reflections in the room, which can introduce phase and amplitude distortions and confuse the soundstage. The biggest culprits are walls, windows, and mirrors. In addition, parallel surfaces (opposite walls, ceilings, and exposed hardwood floors) can lead to standing waves, which greatly affect the frequency response you hear. If possible, install acoustic treatment to minimize these problems.

Keep the cable runs from the amplifier to the monitors as short as possible to prevent induced noise and signal loss. Use fairly large speaker cable, at least 10 or 12 gauge. If you place a speaker near a video monitor, the image might become distorted because of the strong magnetic field from the speaker. Some speakers are magnetically shielded, which prevents this video distortion. In any case, don't put magnetic storage media, such as floppy disks and recording tape, near a speaker; that can spell disaster.

Finally, try not to mix at extremely high levels. This reduces the amount of time you can spend in front of your monitors, thus reducing your chances of getting an accurate mix; it also can permanently damage your hearing. With the proper caution and application, your close-field reference monitors will provide years of useful service, helping you create the best possible sound for your music.

—Scott Wilkinson

[Excerpted from *Anatomy of a Home Studio*, published by EMBooks, an imprint of artistpro.com, LLC.]

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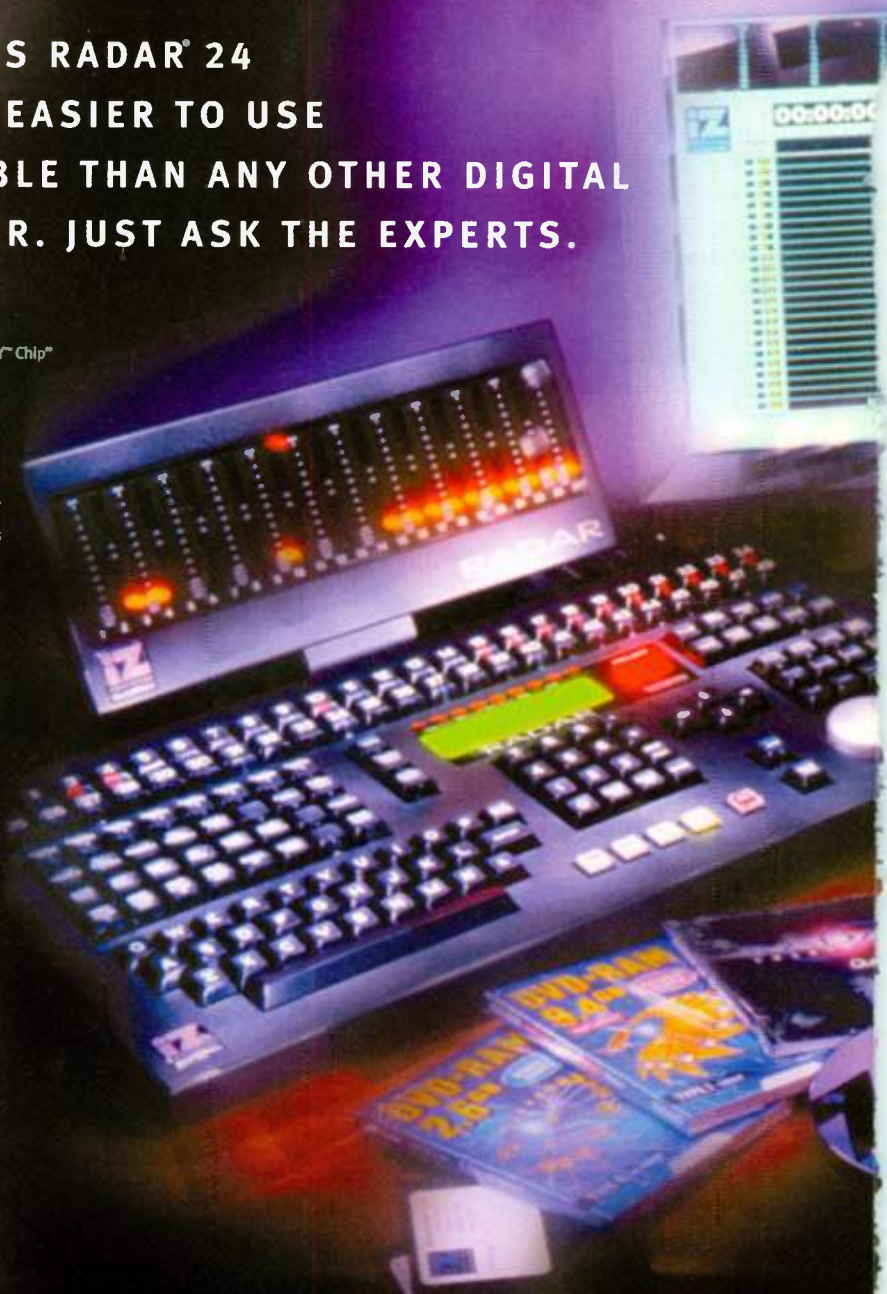

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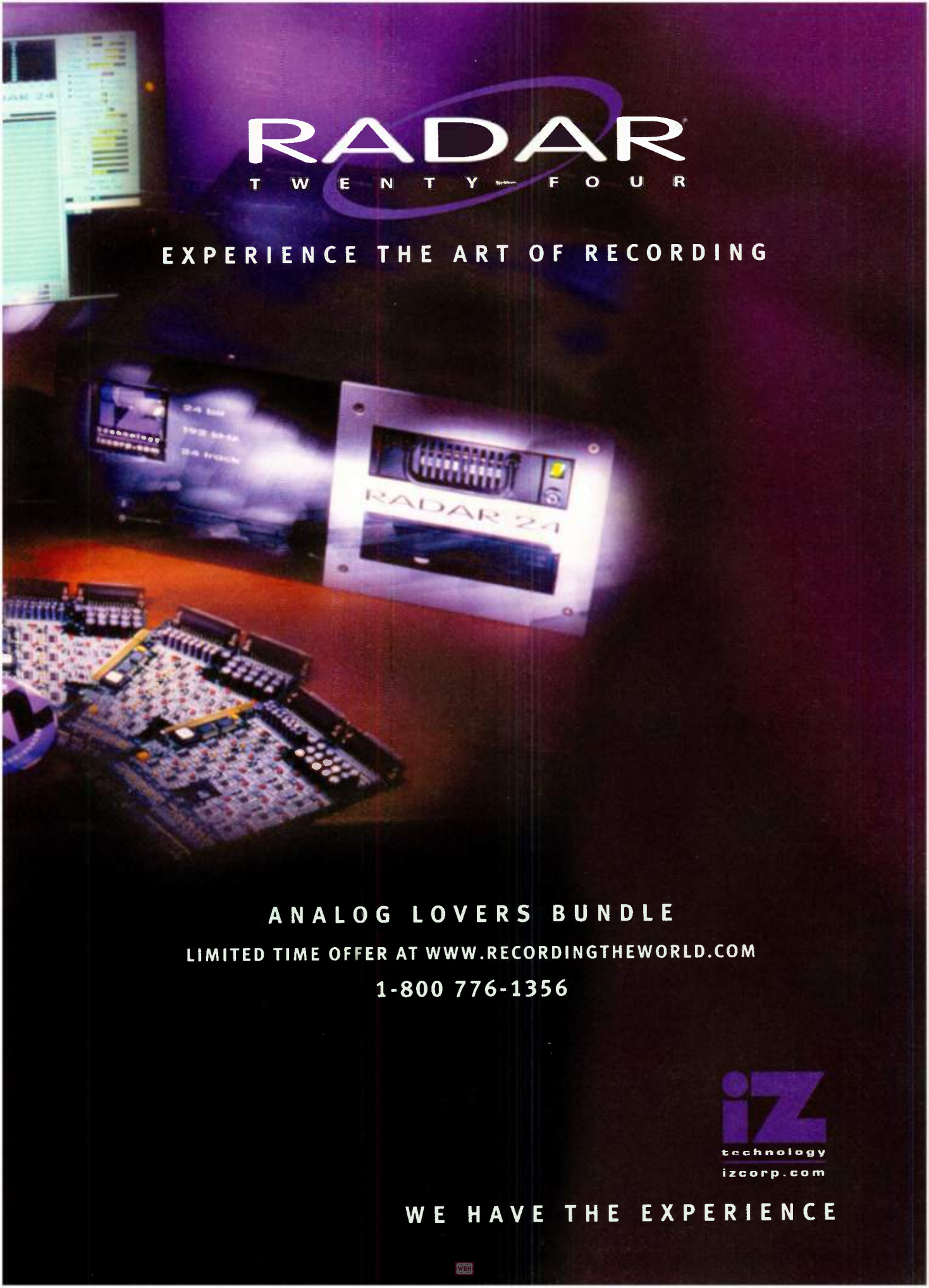
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Four pairs of compact monitors are about the most you can practically compare at one time. One helpful trick is to set up the monitors in an ABCD/ABCD pattern. That way, when listening in stereo, you can simply move a bit to the right or left to get in the sweet spot for each speaker pair: AA, BB, CC or DD.

HEARING IS BELIEVING

Based on this article, you should now have a good idea what to listen for. Remember, you're looking for the monitor that most accurately reproduces what's on your CDs, not the one that makes them sound the best.

In the mono portion of your test, listen for frequency response, timbre, and overall quality. Are there any peaks or dips in the spectrum or any areas that seem to jump out or that you can't hear well? (Your listening session is a good opportunity to correlate frequency response plots with what you hear.) Do the extreme highs sound clear and smooth or are they harsh and biting? Is the bass smooth and tight or does it sound flabby on some notes? Also, how transparent is the sound? Can you clearly hear all the instruments? How about the low-level elements such as harmony vocals, reverb, ambience, and light percussion? Pay particular attention to the midrange, making sure it is clear and revealing rather than soft, muffled, or boxy sounding.

In stereo, listen for the soundstage and try to "see" it in your mind. Instead of seeming to come from the separate speakers, the sound should now fuse into a "stage" that appears to float between and around the speakers (this depends on the mix, too, of course). Close your eyes and listen to the imaging. Are the instruments clearly defined (spatially) on the soundstage?

(This is where having some of your own mixes is helpful, so that you know exactly where and how things are panned.) Also, move off-axis and check the response. Does the tonal balance change drastically as you move your head from side to side, or does it remain about the same? That is, is the sweet spot wide and forgiving, or is it so narrow that you must sit, Sphinxlike, with your head fixed in one position? Also, check out how the monitors sound from across the room. Does the tonal balance change dramatically as you get further away from the speakers?

It is also important to listen at different volumes. Choose three repeatable settings—one quiet, one medium, and one loud—and listen to determine whether the tonal balance remains consistent at all three. At loud levels, listen especially for compression and shifting dynamics, which are not desirable.

Make sure, of course, that you subject each pair of monitors to the same tests, so that the evaluation is thorough and fair. Bring pen and paper as well so you can take notes—there is plenty to keep track of during the test process, and it's easy to forget or confuse your impressions. To keep things focused, you may even want to make a list of the sequence of test events in advance.

Keep your listening session relatively short, preferably no longer than an hour and a half. Even after just one hour of concentrated listening, it can be difficult to focus and make fine distinctions. It is definitely better to come back for a second session than to burn out and lose all perspective on the first day.

If you get down to two favorite monitors and can't decide between them, ask if you can purchase both and then bring one back later for a refund. That will give you a chance to take the monitors home and test them where it really matters—in your own studio. Then you can subject them to an additional battery of tests, including the most important ones, tracking and mixing.

THANKS FOR THE REFERENCE

No matter what monitors you end up with, remember that getting to know them is at least half the game. If a pair of "new and improved" monitors comes along soon after you purchase yours (which is likely), resist the temptation to trade up and focus instead on getting more familiar with what you have. Reference monitors are a tool, not a status symbol. The goal is to learn how to craft mixes that sound great anywhere, no matter what they're played back on. To that end, nothing matters more than knowing your monitors.

Brian Knave *is an associate editor at* **EM.**
Thanks to George Petersen, Scott Wilkinson, Jeff Baust, Dennis Miller, and Bob Hodas.

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Battling Medusa

Wiring your studio like professionals do saves you time and trouble.

If you're like many small-studio owners, the mere thought of venturing behind your gear racks fills you with dread. The tangled mass of cables back there can be so awful and overwhelming that just a glimpse of it will paralyze you.

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ILLUSTRATION BY COURT PATTON

I called upon three studio experts with extensive wiring experience to provide a broad perspective on the professional approach to studio wiring. Ann Dentel, formerly of David Carroll Electronics, spent a lot of time under the floors at Skywalker Sound in Marin County, California, and designed the wiring for numerous studio facilities around the Bay Area. Dentel operates Anzrad Cables, also in the Bay Area, and teaches the art of wiring at Ex'pression Center for New Media in Emeryville, California. I also contacted sound designer and **EM** contributing editor Larry Oppenheimer of Toys in the Attic, a music and audio services company. He has done wiring work at Fantasy Studios, Russian Hill, LucasArts, and Pro Media, all in the Bay Area, in addition to many project-studio installations. Lastly, I spoke with technical wizard Lawrence Fellows-Mannion of Rance Electronics, an Oakland, California–based, full-service recording-studio maintenance and installation business.

BY KAREN STACKPOLE

THE DRAWING BOARD

You've heard the adage "If you fail to plan, then plan to fail." Those are fine words to live by when you take on the task of wiring your studio. But before you dive in headfirst, take stock of your situation and do some serious in-depth analysis. What equipment do you have? How will you use it? What is the studio's layout? Now is the time to decide what setup best suits your working style, ergonomically and in terms of efficiency. Figure out what gear goes into which racks. The pros organize their studios by area, designating locations for specific stations, such as the mixing console, the patch rack, the synth rack, and the computer (see **Fig. 1**).

Once you draw a diagram of where everything will go, the real work can begin. Larry the O suggests that you make an equipment database, documenting every connection in detail. As you separate things by station, generate reports of the inputs and outputs of every piece of gear in a rack, to sort out your cable and connector needs. At the least, draw up an organized, detailed list of each station's gear and connections (see **Fig. 2**). From there, you can figure out what to bring to a patch bay for easy access.

Use the I/O report to make a wire list based on your studio layout. Determine the cables you need and measure the distances those cables must run. Add an extra three to four feet to that for slack to "dress in" your racks, and leave an adequate *service loop* (several feet of slack) that will give you access to the back of your gear later on. Compile a complete supply list that includes cables, patch bays, connectors, and materials for conduits or troughs, according to your master plan. Don't forget to include a cable-management system, incorporating permanent and temporary cable ties, a labeling kit, and other incidentals.

Formulate a budget for high-quality cables, connectors, and patch bays if you want your wiring efforts to last. If you need to economize, don't compromise the grade of components you buy; rather, decide what can be left out of the patch bay, such as mic lines or connections to gear you use for only one purpose. By hardwiring those items, you save on the extra cable required to bring things to a patch bay.

Plan for expansion so that you don't wire yourself into a corner. Fellows-Mannion recommends starting with 20 to 30 percent more cables than you need for a new installation, because you will grow into it; the extra cables will ease the addition of new equipment.

Before you start wiring, thoroughly research grounding, digital routing, and interfacing balanced and unbalanced analog connections. For grounding advice, see "Square One: Power to the People" in the October 1997 issue. For digital-audio etiquette, read "The Route Less Traveled" in the December 2000 issue. Those are important topics that go beyond the scope of this article.

There are rules for interfacing balanced and unbalanced gear, but the most important one is to be consistent. Decide on sensible grounding and interfacing schemes and be consistent as you wire your studio. Stick to that modus operandi religiously to avoid problems later.

CABLES AND SNAKES

Your wiring job should ideally be a one-time affair. At the onset, avoid setting yourself up for failed connections. A sound investment in good cable is a great place to start. You will hear a difference in fidelity, and your cable will last. Because of its performance and affordability, Larry the O prefers cable by Proco and Whirlwind for reliable, everyday needs. High-end cables from companies such as Mogami, Canare, Gotham, and Monster Cable are excellent choices, but from an economic standpoint, it might not make sense to wire your entire studio with the high-end stuff. However, consider using it for some critical connections, such as mix outputs and mic lines. (For more about cable types, see "Good Connections" in the January 2001 issue.)

Fellows-Mannion has this advice about selecting cable for a pro audio-wiring installation: "If you need to get as many lines as possible into a narrow trough or space, use foil-shield cable such as Canare, Belden, or Gepco small gauge, because each pair is smaller. Spiral-wrap cable from Canare, Mogami, or Gotham is best when extra flexibility is needed. Braided-shield cable is by far the best

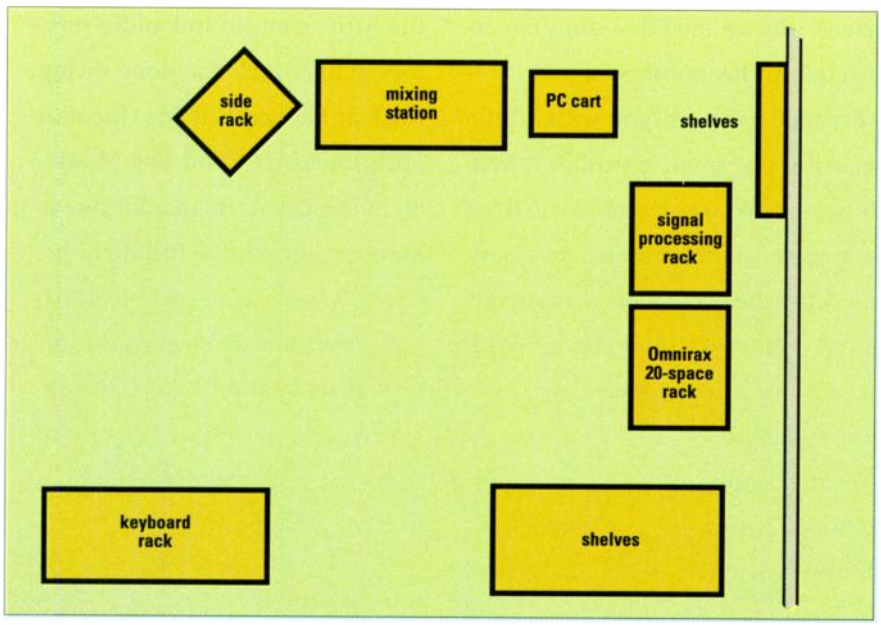

FIG. 1: Separate your studio by area, designating locations for specific stations. This floor plan shows the layout of Larry the O's studio, Toys in the Attic.

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choice for areas where it will be flexed or stepped on a lot. That type of shielding is the most robust of the three cable types."

In addition, Fellows-Mannion recommends using shielded twisted-pair cable whether the gear is balanced or unbalanced. The shield protects against electric fields, and the wire's twisted configuration inside the cable makes it less susceptible to magnetic fields. "Balanced line connections provide some measure of noise rejection," Fellows-Mannion says. "When you have a mile of cable crisscrossing through your studio, you want as much protection as you can get."

Utilize snake cabling as much as possible; it's neater and easier to lay. Larry the O recommends the David Carroll Electronics modular standard of running everything in 8-channel chunks. That makes it easier to change or add things.

"On the other hand, there are advantages to breaking it out," Dentel says. "That way you have a separate cable for each piece of gear, which is easier to dress into the rack and easier to troubleshoot." She admits that separate cables are harder to deal with on the patch bay end because you wind up with a prodigious amount of 4-pair or 2-pair cable coming into the back of your rack, which is more challenging to keep neat.

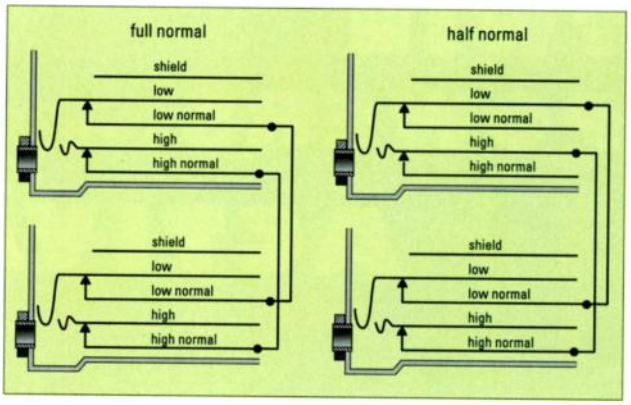


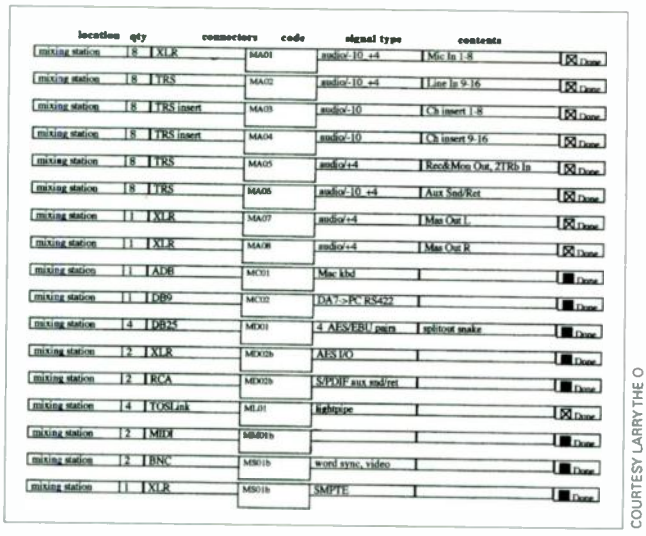
FIG. 3: On the left is a full-normaled connection and on the right, a half-normaled connection.

CABLE LENGTHS

"It's best to make your cables prior to your installation and test each cable before you lay it in and connect it," Larry the O says. "If you don't, you'll have to check every one of your connections after installation. That's less than ideal because it's unlikely that you'll have 100 percent success the first time through." However, Dentel advises that you wait to terminate cables that will be pulled through a conduit or small holes until they have been drawn through and are securely in place.

If you have a movable rack that will be shuffled back and forth, leave a service loop so that you can move the rack as needed. It's also a good idea to leave a service loop at the patch bay so you can work on it (make new connections, fix broken ones, and so forth) if needed. Don't make your cables too short. Remember that tying cables to a structure, such as a rack or table leg, takes extra length. It is better to err on the side of making your cables too long than making them barely long enough.

After you figure out your cable lengths, it's time to acquire the materials. You will need plastic cable ties long enough to handle the girth of the cable bundles you plan to tether, and removable and reusable cable ties, which are often made of Velcro. (You can purchase both types of cable ties inexpensively at most major hardware stores, and better-quality Velcro cable ties are available at electronics stores.) Velcro strips work fine for temporary snake bundling.



location	qty	connectors	code	signal type	contents	
mixing station	8	XLR	MA01	audio/-10, +4	Mic In 1-8	Done
mixing station	8	TRS	MA02	audio/-10, +4	Line In 9-16	Done
mixing station	8	TRS insert	MA03	audio/-10	Ch insert 1-8	Done
mixing station	8	TRS insert	MA04	audio/-10	Ch insert 9-16	Done
mixing station	8	TRS	MA05	audio/+4	Rec&Mon Out, 2TRb In	Done
mixing station	8	TRS	MA06	audio/-10, +4	Aux Snd/Ret	Done
mixing station	1	XLR	MA07	audio/+4	Mas Out L	Done
mixing station	1	XLR	MA08	audio/+4	Mas Out R	Done
mixing station	1	ADB	MC01	Mac kbd		Done
mixing station	1	DB9	MC02	DA7->PC RS422		Done
mixing station	4	DB25	MD01	4 AES/EBU pairs	splitout snake	Done
mixing station	2	XLR	MD02b	AES I/O		Done
mixing station	2	RCA	MD02b	S/PDIF aux snd/ret		Done
mixing station	4	TOSLink	ML01	lightpipe		Done
mixing station	2	MIDI	MM01b			Done
mixing station	2	BNC	MS01b	word sync, video		Done
mixing station	1	XLR	MS01b	SMPTE		Done

COURTESY LARRY THE O

FIG. 2: Compile a list of studio connections, such as this one provided by Larry the O.

DOCK OF THE BAY

The heart of your studio's infrastructure is the patch bay, which lets you bring connections from many gear pieces to one easy-access panel for routing and patching purposes. Consumer-level, 2-conductor, ¼-inch phone-plug patch bays may seem affordable and convenient, but the professionals advise steering clear of them. "You get what you pay for," Larry the O says. "Remember, there are a lot of connections, and you're constantly plugging and unplugging, so the quality of your contacts is important. The cheap ones aren't really worth it."

Fellows-Mannion doesn't recommend consumer bays with ¼-inch jacks on both sides and built-in switching mechanisms that control top-to-bottom normaling. Those switches tend to develop reliability problems because of their construction.

Even if most of your gear is unbalanced, always default to 3-conductor patch bays. They not only provide good protection against interference but also enable studio expansion. A robust yet flexible infrastructure is the best approach. If you want a jack-to-jack bay with jacks on the front and back, a long-frame bay (which uses longer plugs than the standard ¼-inch variety) with

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48 TRS jacks might be the way to go.

If you need to fit many connections into a smaller space, tiny telephone (TT) bays have 96 jacks on the front; the back connections are solder tails, punchdown blocks, or connectors such as Elcos or Edacs. "Soldered connections are the best, but that's a fairly permanent installation," says Fellows-Mannion. Remember to plan for expansion and flexibility. Consider which option suits your studio's cabling needs.

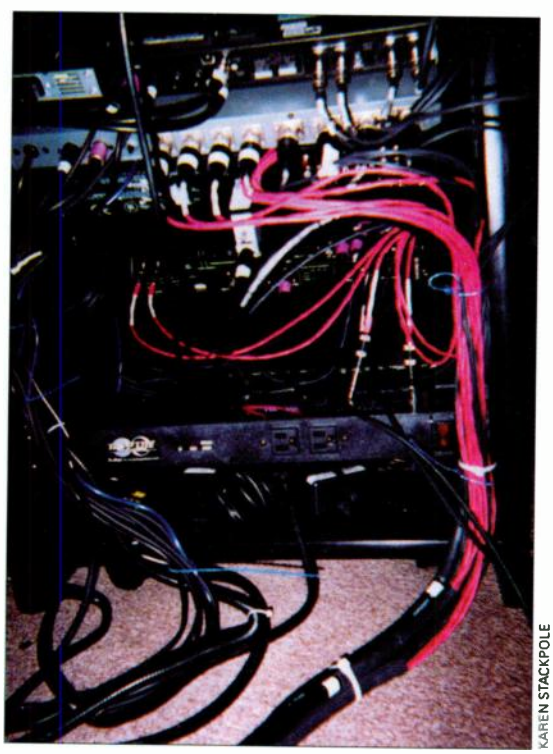
Based on how you work and what gear you routinely use, it's generally a good idea to create "normal" connections between pieces of gear at the patch bay, to make your job more convenient. In this scheme, one device's output, typically in the bay's top row, is connected to the input of another device directly beneath it by default, so you don't need to patch in a cable for normal operation. If you wish to change the setup, the connections are centralized in the patch bay; simply reroute a signal, with a patch cord, to the destination of your choice.

You can make the connections half normal or full normal (see **Fig. 3**). With a half-normal setup, the normal signal path is broken when you plug in to one of the connections, usually the input. With a full normal, the path is broken when you plug in to either connection. Use full-normaling for mic inputs to the board and mic lines from mic panels or snakes, and half-normaling for processors that you might reroute as necessary.

DOCUMENT IT

To make sense of the mess when you need to troubleshoot or add new connections, labeling your cables is crucial. Take your list of cables and connections and develop a code to identify each of them. Be as detailed as possible, indicating the station, function, and line number of each. Larry the O suggests getting a labeler that can print very small type; you can find one at office-supply stores for about $60. He also recommends buying a few extra rolls of labeling tape, because you're going to need them.

"Place labels close to the connector but not too close, in case you need to open the connector," he says. Dentel suggests, as a standard, placing labels about two inches down from each connector. You might have to turn some labels lengthwise on the cable but be consistent. Develop a system and stick to it. Label both ends of the cable and the connectors as well. Keep detailed documentation for reference.



KAREN STACKPOLE

FIG. 4: Keep analog-audio cables separate from digital-audio and AC cables, and tie each bundle to the rack's sides. This rack, at Toys in the Attic, illustrates the arrangement.

LAYING IT ALL OUT

Now that you're ready to lay it all out, which cables can run together and which should be kept separate? First and foremost, keep the AC power cords away from the audio cables so you won't induce noise into the audio. If your audio and AC cables must intersect, cross them at a 90-degree angle. Larry the O prefers to keep digital- and analog-audio cables separate because analog audio is susceptible to picking up noise. He recommends keeping the AC, digital-audio, and analog-audio runs separate, but he says that you can often get away with running digital audio with AC at the rack.

KAREN STACKPOLE

FIG. 5: In the Ex'pression Center for New Media's central machine room, managed cables and bundles cascade down a chute from a raceway in the ceiling to a trough beneath the floor.

Most AC inputs are generally on one side of the equipment rack, so run the AC cords up the side of the rack closest to the most AC inputs. "Unshielded transformers, wall warts, and lump-in-the-line power supplies tend to be big sources of radiated electromagnetic fields," Fellows-Mannion says. "It's a good idea to keep your audio cables away from those. You should also keep your phone lines [telephone, modem, and DSL] away from your audio cables, because ring voltages in phone lines can reach 90V."

However, DC lines from wall warts can run alongside audio cables if necessary, along with other low-voltage power-supply lines such as a DC line running from an outboard power supply to a piece of equipment. "If you're running separate ground wires from your gear to 'star ground' [a grounding scheme], bundle those with your audio

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to keep the loop area between the audio and ground lines small. That helps reduce the possibility of hums and buzzes," Fellows-Mannion says. "Those ground wires should run with your audio and not your AC."

TIE IT UP, TIE IT DOWN

At this point, it's time to bundle the cables you'll run from station to station. Use permanent ties for cables that will be less likely to require access, such as tape machine sends and returns, tie lines (nonassigned line-level cables that allow equipment in different rooms to be connected as necessary), and mic cables. If you think you will have to change something from time to time or you have gear that you move around a lot, use temporary ties to fasten those cables to the main bundle. That will ensure that you can reach the cables when needed.

KAREN STACKPOLE

FIG. 6: This cable bridge protects cables from foot traffic at Toys in the Attic. Note that separate cable bundles from different stations are joined together, and analog audio passes at a right angle over the digital-audio run.

In preparation, take the cables that go from one station to another, such as from the mixing console to the patch bay, and lay them out on the floor. Larry the O recommends you start from the end that will go to the equipment (as opposed to the end that goes to the patch bay) and line up the connectors so they're even. Bundle the cables with a temporary tie about four inches from the connectors. About eight inches down from that (or whatever length you need for your breakout), put on the first permanent cable tie. Then work your way down to the other end, bundling the cables with permanent ties every 6 to 12 inches. Be consistent with the spacing and finish off the bundle with a temporary tie about four inches from the connectors at the other end. Dentel says not to use cable ties if you're laying cable in a trough or pulling it through a conduit.

Now you can go to the equipment end and plug everything in. Run the bundle neatly to the patch bay, placing it out of foot traffic's way. At each rack, run the analog-audio cables up one side and the AC cables up the other. To secure the cable bundles, attach *cable mounts* (gadgets with slots to accommodate cable ties) to the rack's sides (see **Fig. 4**). This ensures that you don't have a zillion cables running amok across the back of your rack. You need to break cables out from your bundle at various levels to connect them to the appropriate patch point or piece of gear.

Larry the O recommends that when you break out your cables, you use a strain-relief bar, which could be as simple as a sturdy wooden dowel running horizontally across the back of your rack. As you loom the cables out to designated connection points, tie the slack to the bar so that the cables' weight doesn't pull on the connectors and jacks.

Above all, be neat and organized. If you get sloppy with your cables, the back of your rack will look like Medusa on a bad hair day—going around the back to troubleshoot an intermittent connection might leave you stony and immobile, as though you laid eyes on the gorgon herself.

OUT OF SIGHT

In a professional installation, most of the cabling is hidden, either in troughs underneath the floor or in cable raceways or trays along walls or above drop ceilings (see **Fig. 5**). If you designed convenient cable passageways into your room, you can lay your cables neatly out of the way.

There are special considerations if you use a conduit, which is commonly made of metal electrical piping. "If you need to put cabling inside of the conduit, it's going to be pulled rather than laid," says Fellows-Mannion. "Manufacturers make special cables for this, with a thicker and stronger jacket designed not to stretch when it is pulled. This is where you have to use single cables and fish tape to pull them together through the conduit." Fish tape is like a long measuring tape that you can use to "fish" the cable through the conduit.

If you don't have a conduit, a raised floor with a cable trough, or a raceway above your ceiling, your cables have to snake across the floor. If it's impossible to keep the cables out of foot traffic, use a cable bridge so that they aren't trampled. "I recommend getting [adequately sized] PVC tubing," says Larry the O. "Saw it in half lengthwise, and place it over the top of the cables. You get a very solid, weight-bearing plastic protector." The main point is that the more out of the way, out of sight, and protected your cables are, the better off you are (see **Fig. 6**).

EXPANDING WORLD

Each wiring installation has its own unique requirements, and you can take many routes with your studio, but several philosophies remain consistent, regardless of the situation. Planning is key. It's important that you don't underestimate factors such as cable lengths, conduit or trough size, and your studio's growth potential. Consistency and neatness are paramount. You must implement a uniform and clear labeling system. Documentation is essential if you don't want to have a nervous breakdown when it's time to troubleshoot.

Do your research about grounding and interfacing balanced and unbalanced gear, as well as analog and digital connections. Adopt a grounding scheme and an interfacing system and be consistent with them. Leave yourself some room to grow. You have carte blanche to be obsessive-compulsive when it comes to wiring your studio. Embrace your inner control freak. Whatever you do, don't let Medusa do you in.

Karen Stackpole *is director of studio maintenance at Ex'pression Center for New Media in Emeryville, California.* *Many thanks to Ann Dentel, Larry the O, and Lawrence Fellows-Mannion for their experiential wiring wisdom.*

We welcome your feedback. E-mail us at emeditorial@intertec.com.

Illustration by Robert Sauber

Master CLASS

TAMING THE TRITON

Korg Triton owners know that calling the Triton just another synthesizer doesn't do it justice. With all the Triton's functions and features, it's more like vast, uncharted terrain, waiting to be explored. In Triton territory the creative adventurer can travel in any direction and discover limitless musical riches.

Trying to explain all there is to know about the Triton is an exercise in futility. In this article I'll reveal some less-than-obvious maneuvers for you to try at home or in the studio. Pointing you in the right direction will help you save time and get your creative juices flowing in new ways.

Without getting too basic, I'll try to be as clear as possible about what operations to perform. If you need help with the fundamentals, a lot of information is available. Check out the sidebar "Get Linked" if you need somewhere to begin.

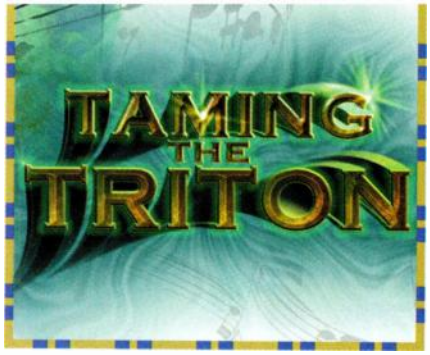
By Clark Salisbury

COSMIC STRING THEORY

Because I'm a guitarist, editing a guitar patch seems like a good starting point. Strap on your favorite pair of rock 'n' roll shoes, and I'll call up program A037: Wet Dist Guitar.

Select the sound and then hit P8: Edit–Insert Effect, or just press Menu and then 8. Turning off the effects makes hearing the edits easier. There's more than one way to disable the effects, but for now, touch the Insert FX tab. You'll find that only one insert effect is being used, 006: OD/Hi.Gain Wah, assigned to IFX1. Touch the on/off switch on the touch screen to turn the effect off. Repeat that process for the Master Effects: hit Menu followed by 9, select the Master FX tab, and turn MFX1 and MFX2 off.

Surveying the terrain of programming technique in Triton territory.



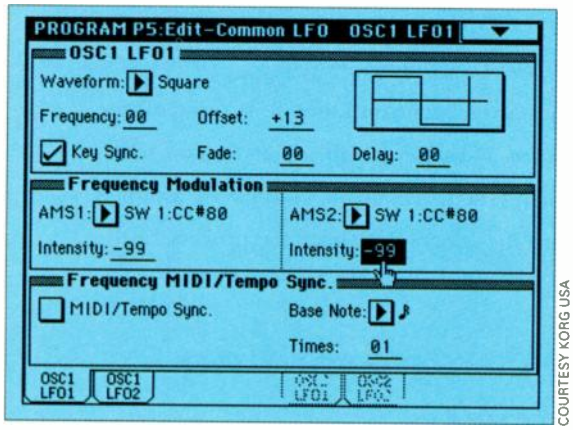
The Korg Triton's depth of programmability ensures that you'll always have plenty of directions to explore. The Triton is available in three keyboard models and a rack-mount version, each with a variety of expansion options.

You can mute the effects from the Global menu, but I prefer to switch them off at the program level. That lets you hear the original sound—with effects—by hitting the Compare key.

Now set up SW1 to toggle between a normal distorted guitar sound and a muted guitar sound. (That function isn't directly supported in the Triton, but don't let that stop you.) To choose the oscillators you'll use, touch Menu, 1, and then the OSC Basic tab. That program is in Double Oscillator mode, with a velocity switch on the first oscillator.

You can leave Oscillator 1 as it is, but select a muted guitar wave for the second oscillator. Press OSC Basic and select ROM wave 151: Dist.Gtr-Mute as the high OSC2 multisample. Because no velocity switching is applied to Oscillator 2, there's no need for concern about the low multisample.

Now set up SW1 to control which oscillator will sound. First turn off the pitch shift that occurs when you hit SW1. Hit Menu and then 2 to move to P2: Edit–Pitch. Select the OSC1 P.Mod tab and set the LFO 2 AMS source to off, or make sure its intensity is set to +00.00.



COURTESY KORG USA

FIG. 1: For an excruciatingly slow LFO, assign a modulator such as SW1 to both of the LFO's Frequency Modulation inputs.

You may wonder what LFO2 has to do with the pitch transpose effect assigned to SW1—a sneaky bit of programming that can come in handy. LFO2 has been assigned a square wave that produces a constant positive output followed by an abrupt change to a negative output, which also remains constant until the abrupt shift back to positive. Hit Menu, 5, and then the OSC1 LFO2 tab to look at that setup.

Normally a square wave might produce a trill effect, but LFO2 is cycling at its lowest frequency, which is quite slow. The low-frequency oscillator (LFO) takes more than 30 seconds to run through the first half of its cycle and then switch from positive to negative. When the LFO is applied to oscillator pitch and you use SW1 to essentially turn LFO2 on and off, SW1 becomes, in effect, a transpose switch.

You can slow LFO2 down even further, increasing the length of time the pitch remains transposed. Assign SW1: CC#80 to AMS1 and AMS2 in the Frequency Modulation field, and set the Intensity for both to –99 (see **Fig. 1**). That will produce an extremely long LFO cycle; I stopped timing it after three minutes, and it still hadn't completed the first half of its cycle.

You can use the same trick to mute Oscillator 1. Move to P4: Edit–Amp and touch the Amp1 Mod. tab. Set the LFO2 AMS source to SW1: CC#80, and set its intensity to –99. Now play a note and hit SW1. The Oscillator 1 sound should mute, leaving only the muted distortion guitar sound assigned to Oscillator 2. Tap SW1 again, and the sound should return.

THE OL' SWITCHEROO

Next, set up the patch so that the Oscillator 2 sound is muted when SW1 is off, letting you switch between Oscillator 1 and Oscillator 2. You might assume that a variation of the control setup you used for Oscillator 1 works, but it doesn't, because amp modulation is applied as a multiple of initial amp level. If you start with a level of 0 and multiply it by applying LFO2, you can guess what happens—nothing. It doesn't matter by how much you multiply a volume of 0; the result will still be 0.

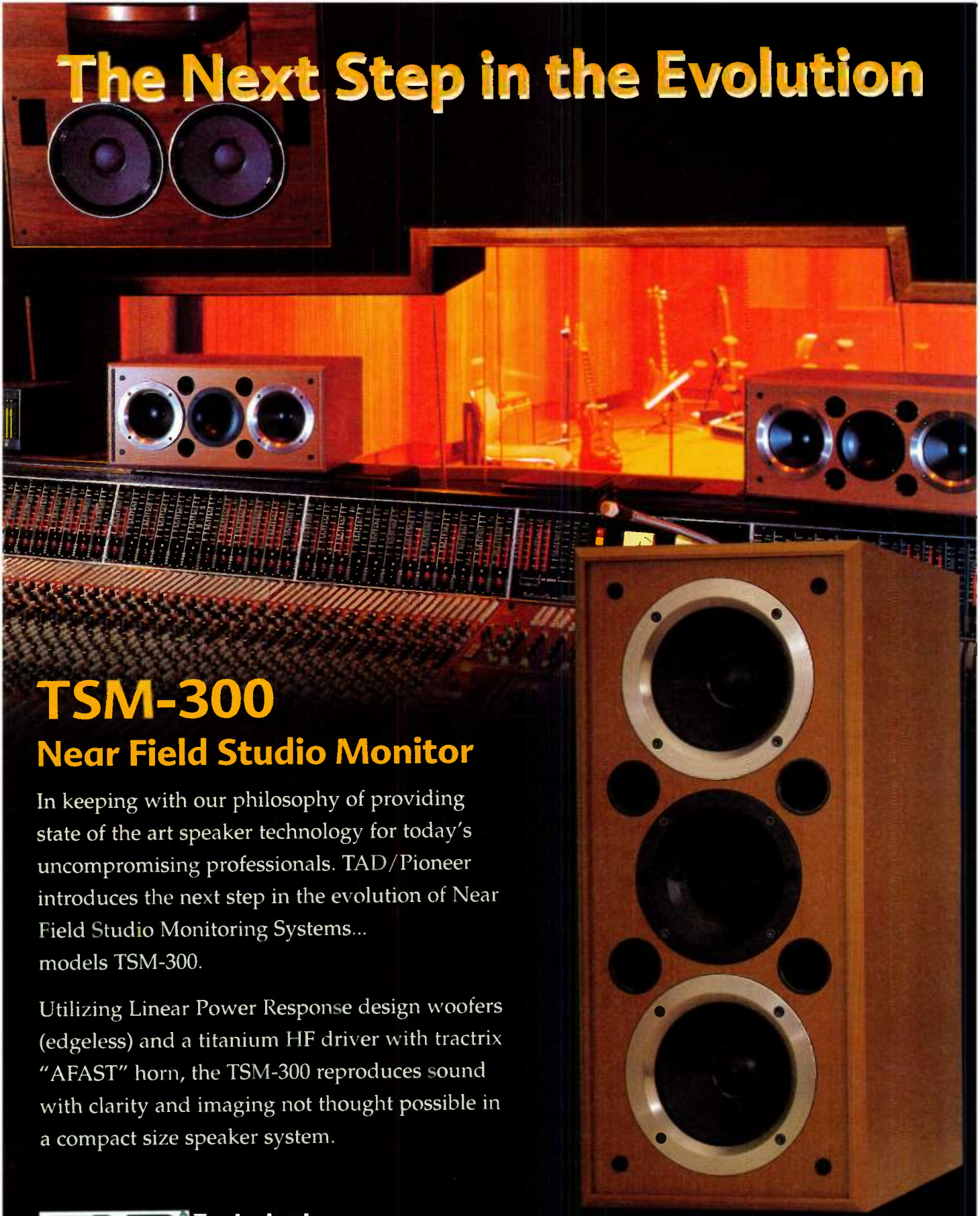
You can, however, use LFO2 to modulate the amp envelope. Press SW1 to mute Oscillator 1. Move to P4: Edit–Amp and press the Amp2 EG tab. From there, you can adjust the envelope so it sounds a bit more like that of a muted guitar. I like these settings:

Level	Start	+99	Attack	+99
	Break	+00	Sustain	+00
Time	Attack	05	Decay	50
	Slope	00	Release	30

They provide a quicker, more mute-like sound. Now get tricky and change the Start and Attack values to +00; leave the other values as they are.

With the Amp envelope levels set to 0, no sound is produced when you play the keyboard. In the screen's Level Modulation section, set AMS to SW1: CC#80, Intensity to +99, Start

The Next Step in the Evolution



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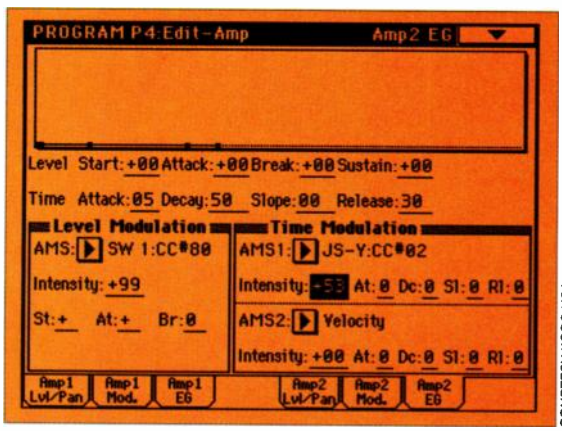
and Attack to +, and leave Break at 0. That drives the Start and Attack levels to +99 when SW1 is pressed, and when you play, you'll hear the muted guitar sound. Turn off SW1 to hear the sustained guitar sound and turn on SW1 to hear the muted guitar sound (see **Fig. 2**).

Wondering why you went through all that to drive the Oscillator 1 Amp level with LFO2 when you could have used SW1 directly to modulate the Amp envelope levels, as you did with Oscillator 2? The reason is apparent if you try it: there's no control input for the Amp envelope sustain level. Either you have to set sustain to 0, causing your guitar sound to eventually decay, or you have to leave sustain set to some positive value. Unfortunately, that causes the sustaining sound to eventually fade back in if you hold the keys down when you play the muted sound. The method I chose might seem a bit roundabout, but it produces the best result.

Finally, if you prefer to have the mute function available only when you hold down SW1, change SW1 from Toggle to Momentary at the Controller tab of P1: Edit–Basic. That edit also allows you to use SW1 like a real-time tremolo control: just hold a note or chord on the keyboard and bounce your finger up and down on SW1.



COURTESY KORG USA

FIG. 2: This envelope uses SW1 to modulate oscillator volume.

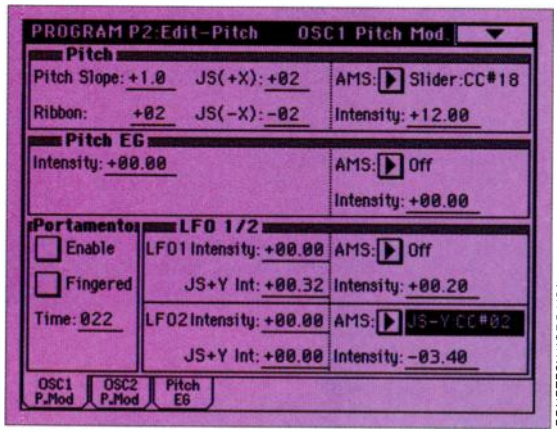
WHAMMY-BAMMY

Next, make a few more changes to the sound of your guitar. With its dual-axis joystick, the Triton is well suited for whammy bar effects. You can use the *x*-axis to control standard bends—whole-step, for example, as in your current patch—and use the *y*-axis to play the virtual guitar's whammy bar. In this case, leave the joystick's positive *y*-axis alone; it can continue to control LFO1 vibrato. Assign the negative *y*-axis to control downward pitch bend.

If you pull the joystick down, it increases Oscillator 1 attack time, so disable that modulation routing. Move to P4: Edit–Amp, touch the Amp1 EG tab, and set AMS2 (in the Time Modulation section) to off. Attack time will remain constant no matter how you move the joystick.

Next, head to P2: Edit–Pitch and touch the OSC1 P.Mod tab. Oops! An Alternate Mod Source (AMS)—the Slider, CC#18—is already assigned to control pitch. You can reassign it, but that's for wimps.

Because LFO2 is already set up as a parameter driver (you have been following along, right?), you can assign the joystick to control the effect on oscillator pitch. For LFO2, set the AMS to JS-Y: CC#02 and set Intensity for the effect you want to produce. For downward bends—analogous to what real whammy bars do—use negative numbers. That sets up the whammy effect for Oscillator 1. To apply the effect to Oscillator 2, as well, replicate the setting for Oscillator 1 (see **Fig. 3**).



COURTESY KORG USA

FIG. 3: Using the joystick to control the ultraslow LFO2 creates an additional pitch-modulation source.

THROUGH A STACK

Here is something I have learned as a guitarist: if you can't get the effect you want out of a stompbox, plug in more stompboxes. With five inserts and two master effects processors, the Triton is jammed full of stompboxes.

For a sound that isn't possible with just one effect, try stacking two or more of the same effect or similar effects. For example, stacking two or more distortion effects makes it possible to produce a much nastier distortion than you can get with a single distortion. Go to P8: Edit–Insert Effect, touch the Insert FX tab, and turn IFX1 back on (you turned it off at the beginning) by pressing the button in the upper-right corner of the IFX1 window. Assign the same effect to IFX2; touch the triangle next to 000: No Effect, hit the Filter/Dynamic tab, select 006: OD/Hi.Gain Wah, and press OK. Touch the Chain checkbox next to IFX1 to pass the signal to IFX 2. Turn IFX2 on and listen to the result (see **Fig. 4**). Pretty gnarly, huh? Now get the effect under control.

Use SW2 to bring the effect in and out, but first switch SW2 from its current function—portamento control—to its basic switch function. Go to P1: Edit–Basic and touch the Controller tab. Select SW2 in the upper-right corner and change it to SW2 Mod.: CC#81. Return to P8: Edit–Insert Effect and select the IFX2 tab.

Change the Wet/Dry mix parameter to Dry. Just to the right, select the source parameter (Src:) and dial up SW2: CC#81. Set the Amount to +100, or less if you don't want so much distortion. When you hit SW2, the extra distortion

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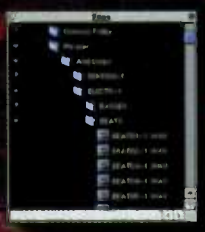

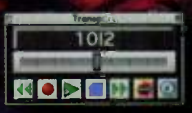

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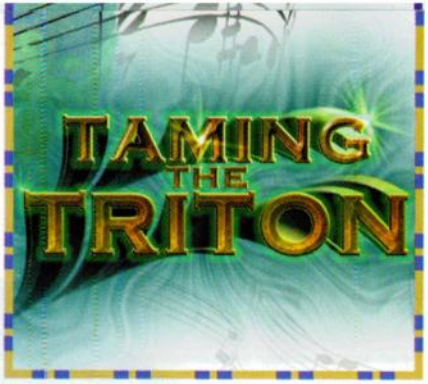


tracks, not just internal sounds. Here's how to make the hookup.

Plug the Triton's main L/R outputs in to a stereo aux return or a pair of channels on your mixer; that serves as your stereo effects return bus. Next, plug at least one of the Triton's individual bus outs in to an input of your mixer. To send stereo sounds to the mixer, you need to use at least two contiguous outputs, such as 1 and 2, or 3 and 4. Then plug a couple of cables from your mixer's stereo effects send or a pair of aux outs in to the Triton's audio inputs. Be careful—with that sort of setup, it's easy to lose track of where you are and bus an aux output back to its input, sending yourself a surprise dose of feedback. I recommend turning all pertinent mixer faders down until the setup is complete and then gently testing everything.

That completes the physical routing of signals (see **Fig. 6**), so move on to the Triton's internal routing. First, touch Global, then Menu followed by 0, and finally the Audio Input tab. At this point route the audio input signal to the effects. In the Input1 and Input2 fields, set the Bus (IFX/Indiv.) Select to L/R; that routes the audio input to the Triton's L/R bus. Set each input Level to 127 and set Pan to C064 (center). Centering the pan lets the effects operate in full stereo.

Finally, in the Input1 field, set Send1 (to MFX1) to 127 and Send2 (to MFX2) to 000. Then do the opposite for Input2 by setting Send1 (to MFX1) to 000 and Send2 (to MFX2) to 127. That sets up each of the Triton's audio inputs to function as a separate effects bus. Input1 is routed to MFX1, and Input2 is routed to MFX2.

By now you should be able to add Triton effects to any of your mixer inputs. To try that out, send a signal to a mixer input—maybe a drum machine or CD, for test purposes—and raise the effects send (or aux send, if that's what you're using) associated with that input. Check out the Triton's input levels by hitting Sampling, selecting the Input Setup tab, and pressing the Rec/Write button. That displays the Triton's input meters, where you can see your input signal level. (The signal from the L/R outs is muted during metering; it comes back when you leave sampling mode.) If the signal is too high or too low, you can use the input trim knob on the back of the Triton to make adjustments, not the recording-level control on the sampling page.

Once you've set the level, press the Combi button and select a combination with a couple of master effects you like. Combination A001: Lonely Moon might make a good starting point because it uses a delay for MFX1 and a reverb for MFX2. Slowly raise the level of whatever you're using for effects returns at the mixer and make sure the Triton's volume slider is up. You should hear the Triton's effects applied to your source material. After you set an appropriate level, listen to the effects with only the first mixer send turned up and then only the second mixer send. If everything has gone according to plan, one send should control the reverb amount and the other should control the delay amount.

Now that you can control the Triton effects levels from your mixer, you can set the master effects bus in the Triton to full on and fully wet. Move to P9: Edit–Master Effect and touch the Master FX tab. Look at Return 1 and Return 2 in the display; if either level is below 127, you get dry signal mixed in with the effected signal. Also pay attention to the chain setup; if the box to the left of the MFX1 and MFX2 fields is checked, one effect's output will be chained into the other, which probably isn't what you want. Touch the MFX1 tab to check the setting for Wet/Dry at the bottom of the page. If that isn't set to Wet, you'll be mixing dry signal in with the effected signal. Be sure to perform that check for MFX2 as well.

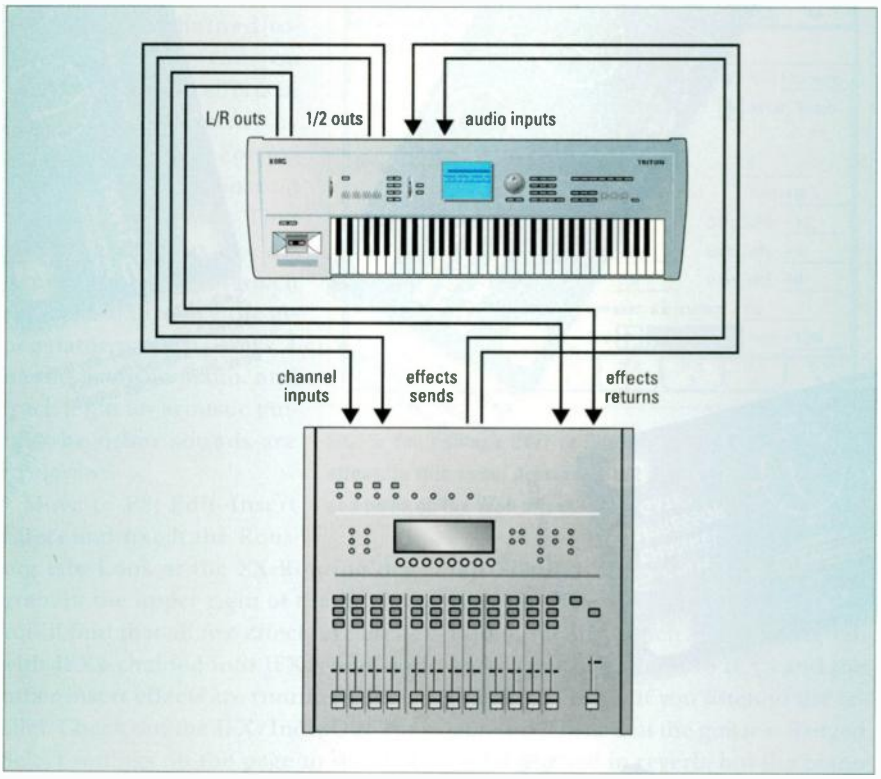
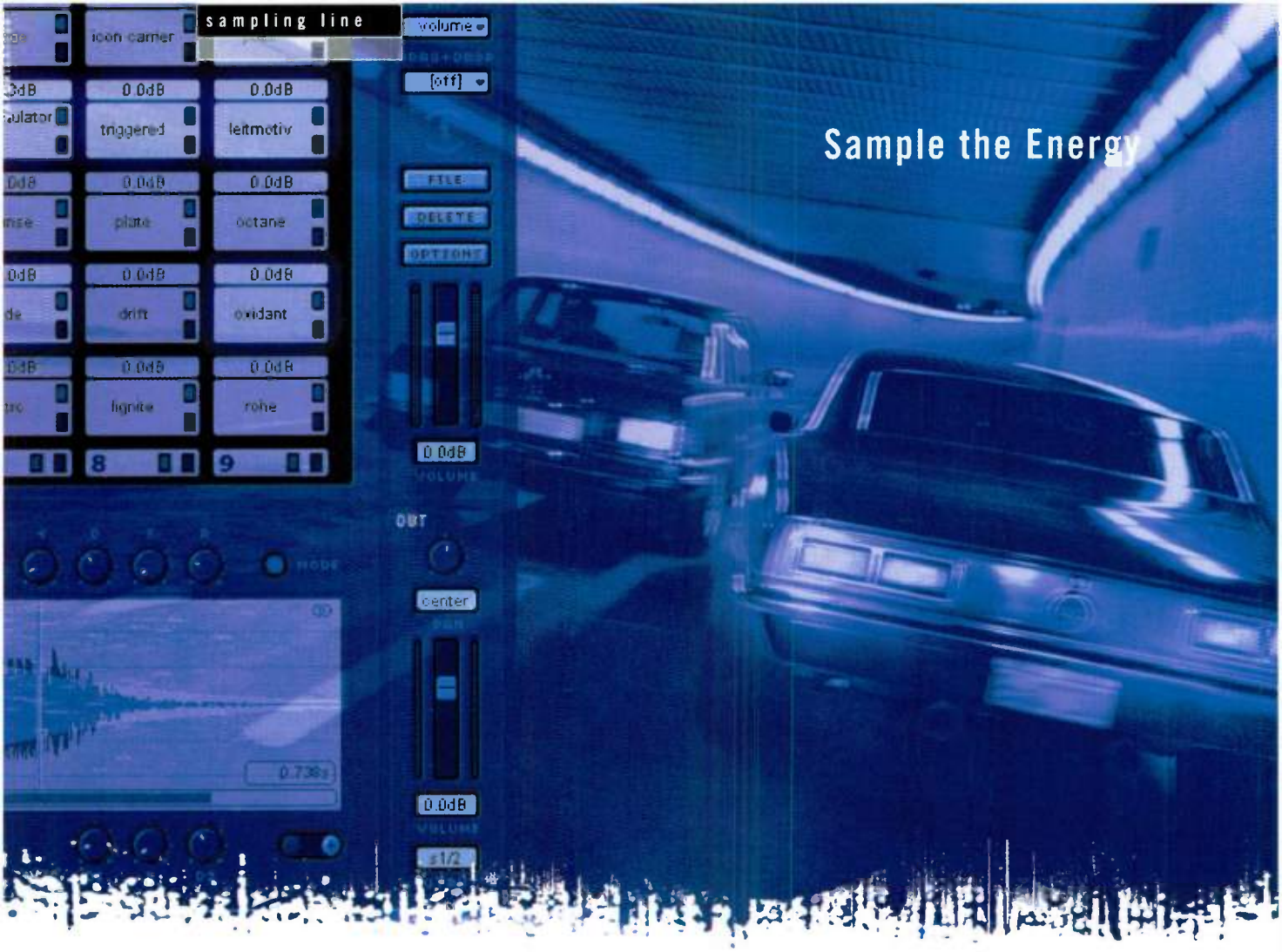
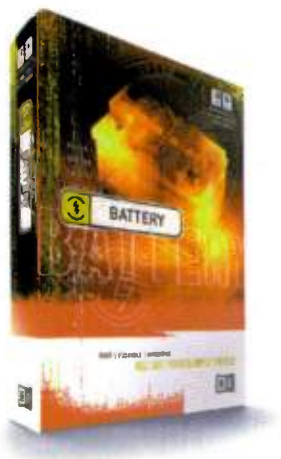


FIG. 6: Who needs a stereo effects processor when a Triton is in the house? This figure illustrates the most flexible signal routing for connecting your Triton to a mixing console.





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TAKE AN ALTERNATE ROUTE

If you try to do any playing or sequencing with the Triton after you make those changes, you will discover that sounds from the Triton appear at the L/R outs, which send them to your mixer's effects return inputs. That problem is easy to remedy—just reassign any wanted Triton sounds to the individual outs.

Reassigning output routings is probably best done from Combi mode; that lets you avoid rewriting individual programs with new output bus settings. You can either send a sound directly to one or two of the individual outs or bus the insert effects to the individual outs. The latter is probably quicker, because it is likely that every sound in a combination will be routed to at least one insert effect. Move to P8: Edit–Insert Effect and touch the Insert FX tab. For each of the insert effects assigned to the L/R output (found in the Bus Sel. column), switch the routing to an individual output or, for stereo sounds, to a pair of outputs. For example, select the tab to the right of the IFX1 field and change it from L/R to 1/2 (you could use output 1 alone, but this wouldn't preserve the stereo image generated by the chorus effect assigned to IFX1). Repeat that procedure for any other insert effects that the combination uses.

Check that none of the other sounds in the combination are routed to the L/R or individual outs. Touch the Routing tab; you'll notice that sounds 6, 7, and 8 are routed to the L/R outs. However, if you check the Play page, you'll see that those sounds are turned off. If any sounds had been routed directly to the L/R outs, you would have wanted to redirect them to the individual outs so they wouldn't be mixed into your effects bus.

Once you've assigned the IFX buses to the individual outputs—as well as any sounds you want to bus directly, bypassing the internal effects—it's a simple matter to run cables from the Triton's individual outputs to your mixer's inputs. To apply the master effects to those sounds, just increase your mixer's effects send for the inputs you're using with the Triton direct outs.

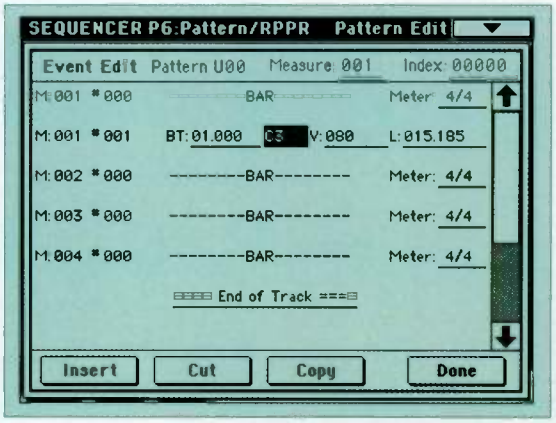

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FIG. 8: With only note data showing, it's easy to find the note you want to cut.

JACK THE RPPR

Listen up, trance fans. The following procedure concerns using the Real-time Pattern Play Record (RPPR) function to generate complex control data. You already know that with the RPPR function you can trigger pattern playback by pressing keys on the keyboard. But those patterns don't have to contain note data; they can just as easily contain sequences of complex controller moves. Recorded controller moves can be applied in real time to any sound you choose. Think of the RPPR as an extremely sophisticated function generator capable of controlling nearly any parameter or combination of parameters you wish.

Press the Seq button, create a new song, and assign a synth sound to the first track. Choose a Program with interesting controller variations—perhaps something such as A008: Rez. Down. Move to P6: Pattern/RPPR and touch the Pattern Edit tab. Either select a blank User Pattern to record or erase an unneeded User Pattern. Touch the pull-down tab in the upper-right corner of the screen and choose Pattern Parameter. The screen that results prompts you to set the length and meter of the pattern. Set the pattern for something not too short—eight measures or so—and leave the meter set to 4/4 (you can try other time signatures later).

Now set the tempo to something not too fast and hit the Rec/Write button. You should hear the metronome start up. When you're ready, hit the Start/Stop button, and after the count off, play and hold a single note. While holding the note, mess around with the controllers: tap your finger up and down on the ribbon, move the joystick around, vary the real-time control knobs, and so on.

When you're done, hit the Start/Stop button to stop recording and touch the RPPR Setup tab. Put a check mark in the Assign box in the RPPR Setup section. In the lower part of the screen are controls for selecting the pattern to assign to any particular key. Assign the pattern but make sure it's the pattern you've been working with. (If you

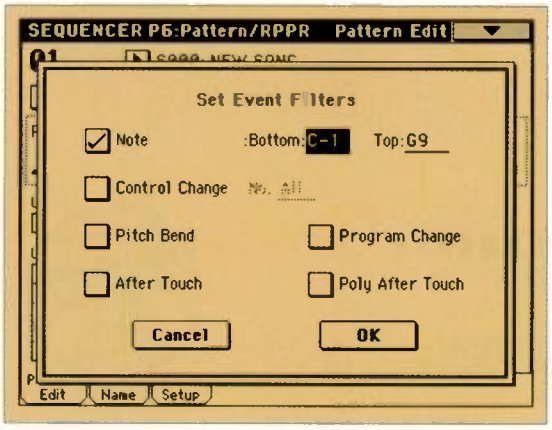

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FIG. 7: To edit only note data, uncheck everything else in the Set Event Filters dialog.

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want to follow my lead, I recorded into User Pattern 01 and assigned it to C♯2 on the keyboard.)

To test your new pattern, go back to P0: Play/Rec. Make sure the RPPR box in the upper right is checked, and play the key that your pattern is assigned to (in my case, it's C♯2). If all of that has gone well, you should hear your pattern play back.

Get ready for the cool part. Go back to P6: Pattern/RPPR, touch the Pattern Edit tab, and select Event Edit from the pull-down menu. In the resulting Set Event Filters dialog box, uncheck everything except Note; you don't want to wade through all the controller data stuffed into that pattern (see **Fig. 7**). Touch OK, and you should see an event list with a single event—the note you played when creating the pattern. Select the event and touch Cut to remove it (see **Fig. 8**). Now touch Done.

Return to P0: Play/Rec and select the track used for the new pattern—in this case track 1. Play a few notes to see that the Triton functions normally. Hold a note or chord and then hit C♯2; that triggers your prerecorded controller data, which is applied to whatever you're playing on the keyboard. In addition, you can change the sound assigned to track 1 and the controller data will be retained, letting you use it with other sounds. Change the sound assigned to track 1 and apply the RPPR-generated controller data to it—cool, huh?

Data you generate with that technique can have all sorts of practical applications. You can create RPPR presets for fades, ultraquick pitch bends (record the data at a slow tempo and then speed it up for playback), and syncopated tremolos and filter sweeps, or you can radically alter a sound instantly or over a long period. You can come up with dozens of applications.

GET LINKED

Triton enthusiasts abound, and lots of them can be found on the Web. Here are a few sites featuring Triton resources, including mailing lists, archives, and shareware applications:

Kevin Goodman's Trinitro Editor-Librarian (Win)
http://magi.com/~goodmank
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The Triton Zone
http://209.35.182.75/trinityhaven/HTML/tritonHOME.htm
Vancesoft Triton Librarian (Win)
http://vancesoft.com/triton

TERRA INCOGNITA

I've only covered a fraction of the things you can coax the Triton into doing, but hopefully I've given you a springboard to launch some of your own ideas. After all, there's still a lot of uncharted territory in Triton country.

Clark Salisbury's *so-called life is based in Portland, Oregon, with an excellent wife and four cats. Special thanks to Jerry Kovarsky at Korg USA.*

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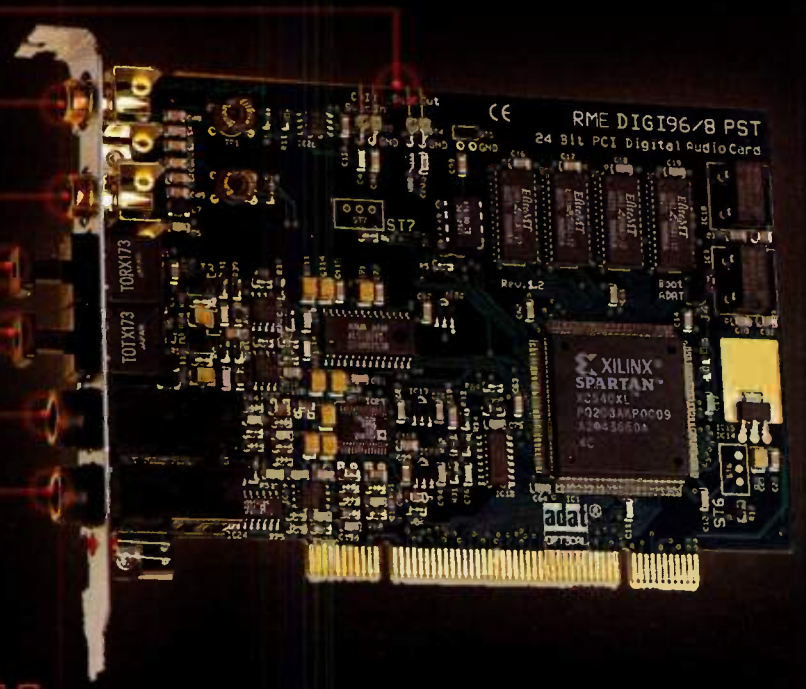

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The Retro ***AS-1*** is extremely powerful software that delivers classic analog synth sounds, mind-warping effects, wicked filters and up to 32 notes of polyphony. ***Voodoo*** rhythm software turns your computer into a programmable beat box that plays samples and standard MIDI files. You can trigger real-time samples and drum patterns from your computer keyboard, MIDI controller, or MIDI application running on the same CPU.

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The ***Studio9000*** is a cutting edge, host-based package with four powerful tools for your creative muse. In this one bundle you get two monophonic, top-shelf analog-modeling synths, an excellent polyphonic sampler, and a kickin' multi-timbral drum machine.

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General MIDI Redux

Explore the bigger musical palette of General MIDI 2.

By Brian Smithers

The birth of MIDI brought a great deal of interconnectivity to the electronic-music world, but within a few years, desktop musicians, multimedia producers, and game developers began clamoring for some level of playback predictability during the exchange of Standard MIDI Files (SMFs). Understandably, composers and arrangers wanted to ensure that piano parts would be played with piano patches and drums wouldn't sound like violins.

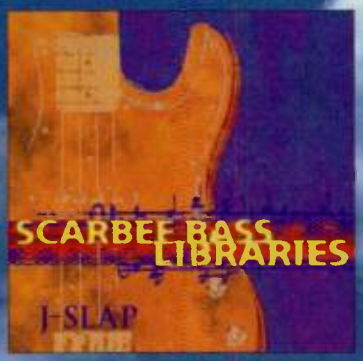
DAVE EMBER

The MIDI Manufacturers Association (MMA) responded by introducing the General MIDI (GM) standard in 1991. General MIDI 1 (or GM1) established a common set of musical criteria for synthesizers and sound modules. The standard was well suited to producing soundtracks for games, distributing MIDI library music, as well as swapping sequences with friends and collaborators.

It wasn't long, however, before musicians and developers demanded more GM instruments with greater control over the instrument sounds. Roland introduced its GS extension to the GM standard, with more patches, more notes in the drum map, and additional elements such as chorus and reverb. Yamaha developed its own XG extension with even more patches and parameters.

In an effort to update the aging GM1 standard, the MMA released General MIDI 2 (GM2) in late 1999. GM2 expands not only the instrument set and drum map but also the number of controller messages a device can implement. To be GM2 compliant, a synth must feature 32-note polyphony, offer a

Room to Play



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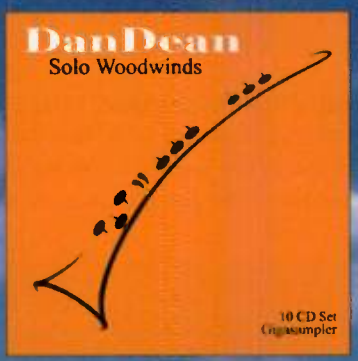

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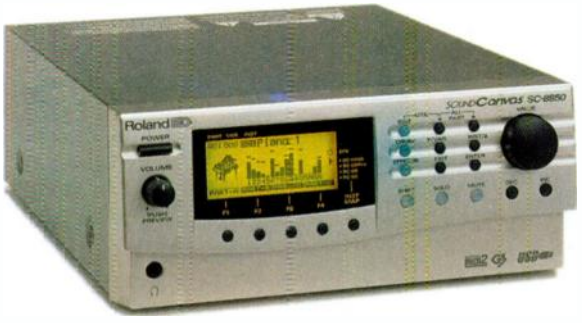
In the February 2001 issue, **EM** examined the crop of GM sound modules; here are some ways to creatively use GM2's newfound power.

GREEDY MUSICIANS

Musicians have long struggled to live within GM1's 24-note polyphony limit, and though the eight additional notes GM2 provides may not seem like much when many instruments boast 128-note polyphony, every little bit helps. Those eight notes can let you use a layered sound that you might have otherwise avoided or add a countermelody that you might have left out.

The new sounds are generally variations of GM1 sounds—for example "wide" and "dark" variations on program 1, Acoustic Grand Piano. In fact, the number of pianos has tripled to include nine acoustic and nine electric pianos. A variety of guitars, basses, string ensembles, and brass patches have also been added. To maintain backward compatibility with GM1 modules, the first sound bank contains the original 128 programs. The new sounds are located in other banks. Anyone familiar with Roland's GS patches will feel at home with the GM2 sound set. (For a closer look, see the sidebar "New Sounds in GM2.")

GM2 devices must include chorus and reverb, which help SMFs sound more polished. Whereas the quality and character of those effects still varies considerably from one device to another, the MMA went so far as to define room size and reverb time, along with specific chorus settings to minimize the variations.



The SC-8850 Sound Canvas from Roland ED provides the user with high-quality GM1, GM2, and GS sound sets in a tabletop unit with a USB connection.

Some of the most interesting developments, though, have to do with the additional control parameters. Some were added to the MIDI language specifically for GM2 purposes.

GENERALLY MARVELOUS

GM2's introduction of Key-Based Instrument Controllers is a major step forward in drum programming. *Key-based* refers to how percussion is mapped, with each MIDI note number triggering a different timbre, such as kick drum on C2 and snare on E2. GM2 lets you map certain controllers to correspond to individual keys and therefore individual drum sounds. That enables composers to do things such as override the default pan positions of drum kit elements. The hi-hat, for example, can be panned all the way to the right instead of centered. Other controllers that can be adjusted per note number include Volume, Reverb Send Level, and Chorus Send Level.

NEW SOUNDS IN GM2

With 87 new instrument timbres to play with, the GM2 sound set offers plenty to like. The piano family has 16 new sounds, most of which are in the electric-piano department. Electric Pianos 1 and 2 each feature a detuned variation and a Velocity mix version; other variations have such names as '60s Electric Piano, EP Legend, and EP Phase. Remember that the specific timbres aren't defined precisely and the sound quality varies greatly according to the price of the synth or sound module. Still, having nine instead of two electric pianos to choose from is bound to make it easier to find the sound you seek.

Chromatic Percussion now includes wide versions of Marimba, Vibraphone, Church Bells, and Carillon. The Organ family has more than doubled, with most of the additions in the Drawbar and Percussive designations. New 12-String and Pedal Steel variations contribute to a tripling of Guitar sounds. The expanded guitar family also includes a Velocity-switched muted Electric Distorted Rhythm Guitar, two new feedback variations, and more.

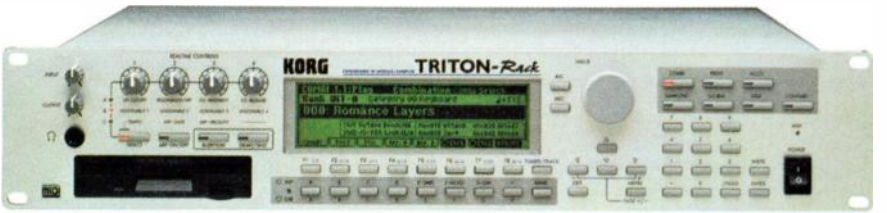
Most new Bass sounds are of the synth-bass variety, including Warm, Resonance, Attack, Rubber, and Clavi types. Strings are almost the same, with only a slow attack violin and Yang Chin added to spice things up, whereas Ensemble sounds are treated more generously. That category adds a patch called Strings and Brass, another called '60s Strings, and a third called Synth Strings. A second Choir Aahs, Humming, and three new Orchestra Hits complete the Ensemble family.

As a multiple-woodwind player, I don't know whether I am relieved or insulted that the Reed and Pipe families are untouched. The Brass family, however, has 11 new sounds, from solo timbres to section sounds and several new synth-brass patches.

Six of the eight new Synth Lead timbres are Lead 1 or 2 variations, including Sine, Sawtooth, Saw + Pulse, and Double Sawtooth sounds. The Synth Pad, Synth SFX, Ethnic Miscellaneous, and Percussive families share 12 new sounds among them, from a Sitar variation to Castanets and Rhythm Box Tom. Although the paucity of new timbres in those categories might disappoint some, the new sound set's focus is clearly on the meat-and-potatoes sounds that are used the most in contemporary music.

It is a common practice for composers to split drum timbres onto different tracks to make it easier to control the balance among the different sounds. For example, because Volume has always been a channel-specific parameter, you couldn't change a snare sound's Volume without affecting the hi-hat. However, if you record them on different tracks (both assigned to channel 10), you can easily change their relative levels by adding a Velocity offset to one track.

Key-Based Instrument Controllers lets you specify the Volume of note number 42 (the hi-hat) independently of the Volume of note number 38 (the snare) using a new Universal Real Time System Exclusive message. The message starts with a header similar to a standard System Exclusive (SysEx) message and then uses a pair of sub-IDs to identify it as a Key-Based Controller message. Next, you define the channel (10 or 11), key (42), controller number (7), and that controller's new value. While you're at it, you can also define new Pan, Chorus, and Reverb messages for the hi-hat within the same message.



In addition to its powerful synthesis and sampling capabilities, Korg's Triton-Rack offers support for both GM1 and GM2.

For that righteous '80s sound, use a key-based Reverb Send Level controller to dial in a bunch of reverb on the snare while keeping the kick drum nice and dry. Add some sparkle to a triangle by increasing its Chorus Send Level without messing up the rest of the kit. (For a closer look at the new GM2 drum sounds, see the sidebar "GM2 Percussion Set.")

CHANGING CHANNELS

Another percussion improvement in GM2 is making channel 11 available as a second Rhythm channel. So much of the music of the last ten years depends on the sound of multiple layered drum kits, yet until now you couldn't even mix and match an Electronic kit hi-hat with Power kit tom-toms, much less stack different kits together.

Having two channels for drums means you can change different channel-specific parameters, such as Volume and Pan, on different instruments. By moving a shaker or tambourine to channel 11 while keeping the rest of the drums on channel 10, you can create some movement by sweeping the shaker around the stereo field using good old controller 10 (Pan).

The Key-Based Instrument Controllers also address some of the benefits of a second Rhythm channel, but they're designed for slightly different purposes. Whereas Key-Based Instrument Controllers are good for rebalancing percussion instruments and for also rearranging the drum kit in the stereo field, their data structure makes them more useful for set-it-and-forget-it use. Using channel 11 for more dynamic effects is easier and eats up less data bandwidth. (Channels 10 and 11 can also be employed as Melody channels for nonpercussion parts; selection is made using a Bank Select message.)

GREATER MANIPULATION

All GM2 devices have a function called the Controller Destination Setting that allows you to assign any combination of six specified parameters to a controller. The setting overrides any default controller assignment. The six parameters are Pitch, Filter Cutoff, Amplitude, LFO Pitch Depth, LFO Filter Depth, and LFO Amplitude Depth. (See the sidebar "Under Control.")

GM2 PERCUSSION SET

So many GM synths have followed the GS/XG example of providing alternate drum kits through program changes that it's easy to forget GM1 specified only one drum kit. Now GM2 canonizes the multiple-kit practice by requiring nine drum kits. With the exception of the SFX Set, they have more similarities than differences, with as few as two notes (both kick drums) separating the Jazz Set from the Standard Set.

The Room Set, for example, offers more resonant toms, and the Power Set replaces the standard toms, kicks, and snares with fat, punchy, reverb-heavy '80s sounds. Two Electronic Sets are included, one of which is called the Analog Set, suggesting sounds modeled after the timbres of the Roland TR-808. Both sets replace the standard toms, kick, and snare with familiar synth versions, and the Analog Set adds cheesy hi-hats and some hand percussion. The Brush Set features the Jazz kick drums and the sounds of brushes tapping, slapping, and swirling on a snare. The Concert Set includes a full octave of timpani, and the SFX Set replaces most Standard timbres with such essential ingredients as laughter, applause, dog bark, and bubble.

The drum map grew as well, adding seven new notes above and eight new notes below the GM1 map. Additional timbres include shaker, muted and open surdo, metronome click, and scratch push and pull. Although each sound should appeal to somebody, one glaring hole in the drum map wasn't plugged this time around. Ignoring XG's shining example, GM2 fails to provide a snare roll, one of the most useful sounds and also one of the most difficult to fake. Oh well, maybe next time.

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Say you want your Lead patch to slide up a half step and wah every time you lean into the keys. If you're brave enough to write a few SysEx lines, you have it nailed.

The Controller Destination Setting syntax starts with the Universal Real Time SysEx header (F0H 7FH), followed by the device ID. That in turn is followed by two sub-ID numbers. The first is 09, the ID for Controller Destination Setting, and the second is 01H for Channel Pressure or 03H for Control Change (CC). The next byte specifies the channel for which you are implementing the assignment. Remember that channels are one of the few things in MIDI that are commonly numbered from one instead of zero, so if you mean channel 5, enter a 04H value. If you are assigning a controller, the next byte is the number of that controller, but for the purpose of our Aftertouch example, you can skip that step.

The next two bytes are for the destination parameter and the range of the parameter's variation. You can map more than one parameter by repeating the destination/range sequence for each additional parameter. In that case, you assign Pitch Control to Channel Pressure with the message 00H and set a 1-semitone range with the message 41H. Follow that with a 01 value to add filter cutoff control to the equation, and set a range of +4,800 cents (four octaves) with a 60H value. Naturally, a bit of experimentation is necessary to determine the range required to produce your wah effect.

The entire message ends with the customary End of Exclusive message, F7H. Once that is done, your lead sound will exhibit the desired bend/wah effect until you assign a different controller destination. You can use the same process to assign a general-purpose controller to filter cutoff, letting you include a filter sweep's classic sound in your GM2 sequence.

GIMME MORE

Several of GM2's more interesting features are, unfortunately, only recommendations, not requirements. From a practical perspective, that means a GM2 synth might not respond to the messages. In the hope that manufacturers will recognize that those features are too cool to ignore, here's a look at some of them.

Filter resonance, which is an emphasis of frequencies near the filter cutoff frequency, adds a certain edge to the sound. CC 71 is defined in the GM2 specification as controlling the resonance degree. When combined with CC 74, Brightness (which controls the filter cutoff) provides plenty of sound-shaping power. Whereas the filter's exact behavior is left to the manufacturer, CC 71 provides a relative change in the resonance effect, increasing the strength of the resonance with values higher than 64 and decreasing the resonance's strength with values lower than 64. Brightness is also a relative control.

One of the biggest shortcomings of GM acoustic-instrument patches (or almost any synthesizer's acoustic emulations, for that matter) is the one-size-fits-all way the notes start and stop. Wind and string players devote their entire lives to articulating notes in precise yet constantly varying ways, but a synthesized trumpet or violin note always starts and stops in a manner defined by a fixed envelope.

Fortunately, CCs 73, 75, and 72 have come to the rescue. They're recommended by the GM2 specification for controlling attack time, decay time, and release time, respectively—three of the four parameters that make up a typical ADSR envelope. Any GM2 synthesizer that implements those CCs lets you create more realistic acoustic sounds.

For example, the distinction between GM programs 49 and 50, the two String Ensembles, has always been that String Ensemble 2 has a slower attack and release than String Ensemble 1. The envelope controllers let you fill in the gaps between those two extremes. By assigning CC 73 to a data wheel or slider on your controller keyboard, you can coax hard or soft attacks (or any degree in between) from a string program without changing patches.

Decay is the envelope's second stage, and a rapid decay (short decay time) makes a note sound more accented. Release time is the length of time from the Note Off message to the actual end of the tone. Plucked strings have a long release time, and staccato brasses have a short release time. All three envelope controllers make relative adjustments to the program's default envelope parameters, with a value of 64 corresponding to the default.

Under Control

Controller Destination Setting is one of several new functions added to the MIDI 1.0 specification for inclusion in General MIDI 2. It lets you assign any of six controllers to Channel Pressure or a CC. The assignment is made as a Universal Real Time SysEx message as follows:

F0H 7FH	Universal Real Time SysEx header
<device ID>	ID of target device
09H	sub-ID#1; Controller Destination Setting
01H	sub-ID#2; Channel Pressure (03H for Control Change)
04H	MIDI channel 5 (channels 1–16 are 00H-0FH) [CC Controller Number if using Control Change]
00H	Pitch Control
41H	Range of 1 semitone
01H	Filter Cutoff
60H	Range of 4 octaves
F7H	End of Exclusive

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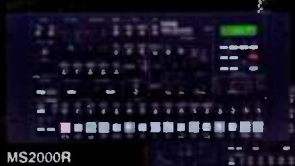

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That is a perfect application for a keyboard with multiple data sliders, such as the Kurzweil K2500, with its eight assignable faders. With each envelope controller assigned to a different fader, even a keyboard-challenged klutz could play more convincing acoustic-instrument articulations.

GAINING MOMENTUM

Now that Yamaha and Roland have agreed to support GM2 and are encouraging others to do so, it's a good bet that compliant devices will be widely available soon. When that happens, you'll be able to write GM2 sequences and be assured that they'll sound reasonably consistent, no matter who plays them back.

Given the enormous number of GM1 devices, however, you might not want to lean on the new features too much. In particular, filter-based effects could come out sounding like static drones on non-GM2 devices. For maximum GM1 compatibility, you might also want to avoid using channel 11 for drums.

Fortunately, the new instrument sounds are in different banks and feature the same program numbers as the sounds they most resemble. Because most GM1 devices simply ignore bank changes, the new sounds play back like their GM1 siblings. In most instances, the results should not be catastrophic.

The GM specification has been enormously successful in bringing compatibility to SMFs. GM2 will prove irresistible to composers and manufacturers for its expanded sound palette and its new sound-shaping tools. In time, people will get greedy again and insist on GM3 or some GS/XG/GM hybrid, but until then there are lots of cool new options to keep everyone busy.

Brian Smithers *is associate course director of MIDI at Full Sail Real World Education. You can reach him through his Web site at http://members.aol.com/notebooks1.*

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The Shocking Truth

The secrets of electricity revealed.

By Scott Wilkinson

During the past 15 years, many fundamental music-technology concepts have been explained in "Square One" (originally titled "From the Top"). In 1997 EM *technical editor Scott Wilkinson combined many of those columns into a comprehensive primer titled* Anatomy of a Home Studio: How Everything Really Works, from Microphones to MIDI, *published by EMBooks, an imprint of Artistpro.com (www.artistpro.com). Our readership has continued to grow, and new readers shouldn't be left behind. Rather than try to reinvent the wheel, we will periodically reprint excerpts from the book as "Square One Classics." These articles will clarify the essential, unchanging concepts that make it possible to be an electronic musician.*


GEORGE ABE

Electricity can be mysterious to many people, even those who have worked with music technology for years. But to get the most from the tools of electronic music, you need to understand the fundamental concepts of electricity. For example, manufacturer specifications mean nothing without them. (Some specs mean nothing anyway, but that's another story.) How can you make an informed purchasing decision without understanding what the specifications mean?

For now, I'll concentrate on four basic properties of electricity: current, voltage, impedance, and power. Understanding these properties is essential if you want to know how electronic equipment works. It's also critical for comprehending other concepts, such as decibels (which I'll explain in the next column).

CURRENT AND VOLTAGE

Most audio signals consist of electrons flowing through a conductor, such as a

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copper wire. This flow of electrons is called a *current*. The amount of current is measured in units called *amperes*, or *amps* (abbreviated *A*), after French physicist Andre Ampere; it is represented by the letter *I* in electrical equations.

There are two types of current: direct and alternating. Direct current (DC) flows steadily in one direction through a conductor; alternating current (AC) changes direction in the conductor at various frequencies. Analog-audio signals are alternating currents with waveforms and frequencies that correspond to acoustic sounds. These audio signals are called analog because the current's waveform is analogous to the acoustic waveform it represents.

An *electromotive force* (EMF) causes current to flow. The name makes sense when you think about it: EMF is a force that makes electrons move. EMF is more commonly called *voltage*, which is measured in units called *volts*, after Italian physicist Alessandro Volta, and is abbreviated *V*. It is represented by the letters *V* or *E* in electrical equations. Voltage is produced in many different ways, such as chemical reactions in a battery.

For a helpful analogy, think of voltage as the height of a hill. Because there is a difference in height between the hill's top and bottom, a ball rolls down the hill under gravity's influence. When the ball is at the top of the hill, it has potential energy; that is, it has the potential to move down. As it rolls down the hill, the potential energy is converted into kinetic energy, the energy of motion.

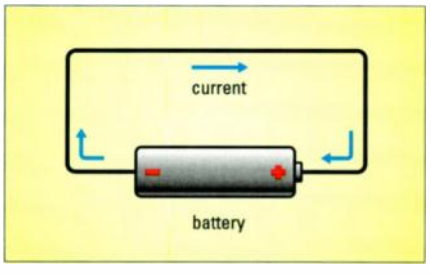

FIG. 1: If you connect a battery's poles with a conductor, current flows from the negative to the positive pole. The amount of current depends on the conductor's impedance.

So the hill's top and bottom are at different heights, and the ball moves from one to the other. The same is true for voltage and current. Any voltage source has two *poles*, and electrons flow from one to the other. There is a potential difference in voltage between these poles: the bigger the difference, the greater the potential for moving electrons. However, that potential can't be fulfilled until an electrical conductor connects the two poles. If you connect the poles in this way, you create a circuit—or closed loop—through which the current flows.

One common voltage source is a battery, which has positive and negative poles. If you connect a conductor to these poles, electrons flow from the negative pole to the positive pole (see **Fig. 1**). If you've played with magnets, you know that opposite poles attract and similar poles repel each other. The same is true for electrons, which are negatively charged. They are repelled by a battery's negative pole and attracted to its positive pole.

AC/DC

Because a battery's voltage produces a direct current, its voltage is specified in units of VDC. If the poles of a voltage source alternate between positive and negative (as they do in a wall's power outlet, for example), the current changes direction periodically, and the voltage is specified in VAC.

Measuring the voltage from a battery or other DC source is easy. Returning to the hill analogy, the higher the hill, the more potential energy the ball has. The battery's voltage is analogous to the height of the hill: the more voltage, the more potential it has for moving electrons. To measure a battery's voltage, simply attach the two leads from a voltmeter to the poles and read the voltage (see **Fig. 2**).

Measuring alternating voltages is not so straightforward. You could simply measure the highest voltage level as it varies up and down, but what if the peak level changes from one cycle to the next, as it does at the output of most audio equipment? Taking the average of several peaks is better, but engineers have devised a more accurate way to measure alternating voltages: *root mean square* (RMS).

FIG. 2: A voltmeter measures the DC or AC voltage between two points in an electrical circuit. Many voltmeters can also gauge DC resistance; such devices are called *volt/ohm meters* (VOMs).

Here's how RMS voltage is calculated. (The process sounds complicated, but don't worry—you never have to do it. I'm describing it to explain what RMS means.) First, measure the instantaneous voltage value at many points during one complete cycle; that is similar to digital-audio recording. Then, square each voltage value (that is, multiply the value by itself). Next, calculate the average of those squared values and take the square root of that average. If the voltage variation takes the form of a sine wave with constant amplitude (as the voltage from a wall outlet does), the calculation becomes simpler: multiply the peak value by 0.707.

This is relatively complicated, but it yields a meaningful voltage value, even in the face of different peak levels

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over time. Fortunately, you don't need to worry about this process; anyone who wants to measure an alternating voltage can simply connect a voltmeter to the poles of the voltage source. The voltmeter does the squaring and averaging, giving you a readout in VRMS or VAC.

IMPEDANCE

In virtually all electrical circuits, there is some opposition to the flow of current; even copper wire opposes it to some degree. (The only exception is a circuit made with superconducting material, which exhibits practically no opposition to current. As of this writing, superconductors only exist inside laboratories.)

The opposition to direct current is called *resistance,* which is measured in units called *ohms,* after German physicist Georg Ohm. Resistance is abbreviated with the Greek letter omega (Ω), and it is represented in electrical equations by the letter *R.*

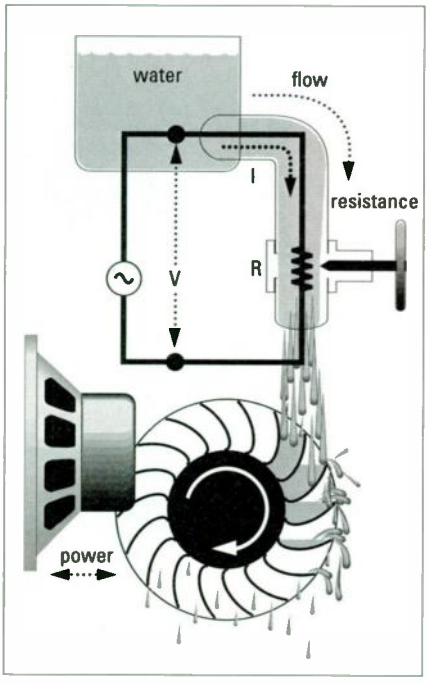


FIG. 3: This mechanical water system is analogous to a simple electrical circuit. The water tank's height above the water wheel corresponds to voltage, and the flow of water through the pipe represents current. The valve offers resistance to the flow of water, which corresponds to impedance, and the wheel turns as the water falls on it, which represents power.

The opposition to alternating current is called *impedance,* which is also measured in ohms, but it's represented in electrical equations by the letter *Z.* Impedance is the sum of DC resistance and the *reactance* of the circuit, also measured in ohms and represented by the letter *X.* (To be completely accurate, reactance includes two parts, capacitive and inductive reactance, but this is not important for now.) Among other things, reactance depends on the alternating current's frequency.

Here's an important thing to understand: a circuit's impedance determines the *load* it places on the voltage source. If the circuit's impedance is high, it doesn't let much current flow, which places little demand on the voltage source to move electrons. Therefore, high impedance puts a small load on the voltage source. However, if impedance is low, the circuit doesn't resist the flow of current, which places greater demand on the voltage source to move electrons. Low impedance places a large load on the source. Impedance and load are inversely related; if impedance is high, the load is small, and vice versa.

In audio connections, you should be aware of two points of impedance: a source device's *output impedance* and a destination device's *input impedance.* In general, the lower the input impedance of the destination device, the greater the source device's load. As a result, a destination device's input impedance should be at least ten times the source device's output impedance.

The output impedance of most professional microphones is low, generally in the range of 150Ω, so mic preamps should have an input impedance of about 1,500Ω or 1.5 kilo-ohms (kΩ). (Some mic preamps have an input impedance as high as 10 kΩ, but the range from 1.5 to 3 kΩ is more typical.) Line-level devices, such as synths, also exhibit low output impedances in the 50 to 100Ω range, and they operate well with any input impedance more than 1 kΩ. Older synths and some consumer hi-fi equipment often have output impedances in the 100Ω to 1 kΩ range, which requires the destination device to have an input impedance in the 1 to 10 kΩ range.

An electric guitar's output impedance depends on the pickup design, the settings of the volume and tone controls, and the frequency produced. When the volume knob is turned up (which is usually the case), the guitar's output impedance is typically 3 to 10 kΩ at low frequencies and 100 to 500 kΩ at 10 kHz. When the volume is down, the output impedance is more constant, but it still varies by a factor of ten from low to high frequencies.

In addition, guitars are very sensitive to the input impedance of an amp or DI box; the higher the input impedance, the better the frequency response. Typical guitar amps have an input impedance in the 1 megaohm (MΩ) range, which gives you a high-frequency response as high as 20 kHz with single-coil or humbucking pickups; low-impedance pickups provide even more high-frequency response.

The relationship between voltage, current, and impedance is defined by *Ohm's Law,* which was derived by Ohm in 1827. The law can be stated in three equivalent ways:

$$V = I \times Z$$
$$I = V/Z$$
$$Z = V/I$$

Among other things, Ohm's Law clarifies the concept of load. Take a look at the first form of the law. If the voltage remains constant, the current will be high if the impedance is low, and vice versa.

POWER

Another common electrical quantity is *power,* which measures how much work can be done by a given voltage and current through a particular impedance. It is represented by the letter *P* in electrical equations, measured in units called watts (after Scottish engineer James Watt), and abbreviated *W* in measurements. DC electrical power is defined by *Joule's Law,* which is named for British physicist James Joule:

$$P = V \times I$$

If voltage and current alternate—as in an audio signal—so does power. As a result, alternating power is often expressed in *watts RMS*. This should be familiar to anyone who has shopped for a power amplifier. Joule's Law is slightly different for AC circuits:

$$P = K \times V \times I$$

K is a constant called the *power factor*, which depends on the circuit's reactance. Its value is always in the range of +1 to −1.

Here's another analogy that illustrates these concepts. Imagine a water tower with a pipe and a valve that lets the water flow from the tank to turn a water wheel (see **Fig. 3**). The distance between the tank and the water wheel corresponds to voltage; the higher the tank above the wheel, the more potential there is for the water to flow. The flow of water through the pipe corresponds to current.

The valve can be opened to different degrees, allowing more or less water through. As you might guess, this corresponds to impedance. The water turns the water wheel, which lets the wheel perform work (say, grinding flour). This corresponds to power. If the valve is mostly closed (impedance is high), little water flows (current is low), and the wheel does little work (power is low). On the other hand, if the valve is mostly open (impedance is low), lots of water flows (current is high), and the wheel can do lots of work (power is high).

The concepts of voltage, current, impedance, and power are essential to understanding basic electrical circuits and specifications. Once those concepts feel familiar, you'll find making the proper connections between pieces of equipment much easier. You'll also be able to make more sense of manufacturer specifications, which should help you make better purchasing decisions.

EM *technical editor* **Scott Wilkinson** *has been zapped more than once after carelessly touching the poles of an AC wall outlet.*

We welcome your feedback. E-mail us at emeditorial@intertec.com.

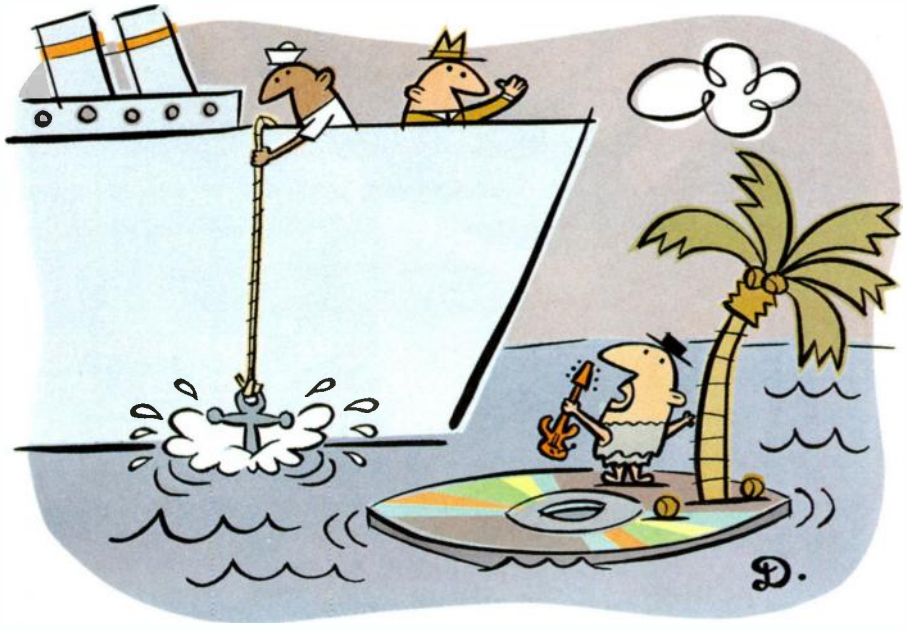
Publish or Perish?

Master the mysteries of publishing—music industry's powerful side.

By Mary Cosola

The music industry's business end is not about art; it is about acquiring and maintaining power—power to make deals, power to make money, and power to make or break stars. For every Clive Davis, David Geffen, and Jimmy Iovine you read about, a hundred lesser-known people are generating the currency that makes the industry go round—the songs. Without songs, there is no music industry. The songs don't hold the power, though; that power lies with the publishers, the people who can get the songs to those who matter.

Music publishing doesn't attract the kind of mainstream press that heads of record companies or famous artists do. For artists just starting out in the business, recording contracts and artist advances are usually familiar concepts, but the ins and outs of music publishing often remain an enigma. Take a look at the following basics of publishing and how they fit in to your plans for your music.



JACK DESROCHER

WHAT IS PUBLISHING?

You are probably familiar with the term *publishing* as it applies to printed materials such as books, magazines, and newsletters: when something is printed and distributed, it is published. That simple concept also applies to song publishing, although a song does not have to be in print to be considered published. It merely needs to be composed and either notated or recorded. However, the simplicity ends there. To understand more about music publishing and its importance, you need to know how it fits in to the music industry's general structure.

Songs generate revenue in many ways, and the two you may be familiar with are *mechanical royalties* and *performance*

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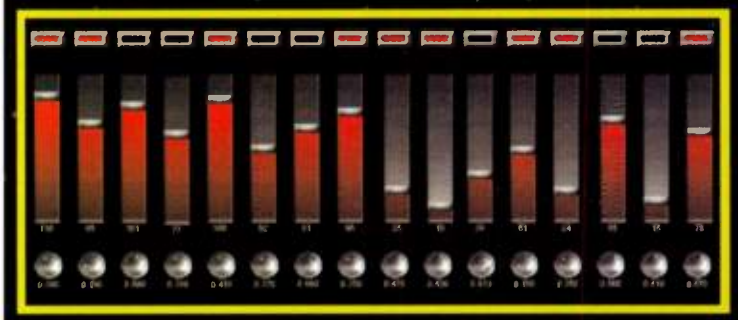

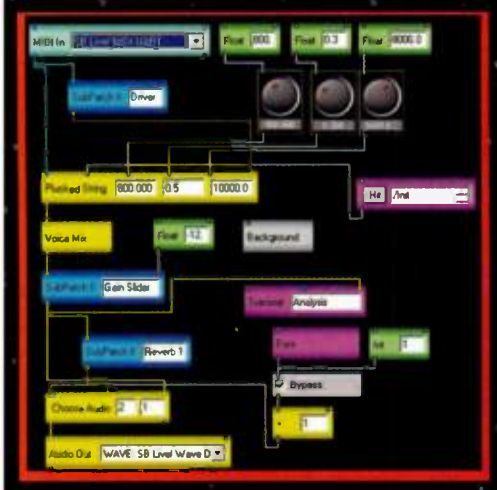

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royalties. Mechanical royalties are paid when a song on a CD or other medium is sold; performance royalties are paid when a song is played on radio or otherwise broadcast—by being played, for example, in a nightclub or a restaurant. Those royalties are split between the songwriter and the publisher, and the percentage each receives depends on the deal the two entities have.

When you sign a publishing deal, you are handing the publisher part ownership of your songs. In exchange for that, the publisher works on getting your songs recorded by major artists, placing songs on soundtracks and in commercials, and finding other projects that need music. That's not to say that you must sign with a publisher. You are perfectly free to act as your own publisher; many artists do. As you will see, though, being your own publisher is not necessarily the easy way out.

FIRST THINGS FIRST

When you start writing and recording songs, you need to address a few simple but crucial legal issues immediately, regardless of who handles your publishing. First, copyright your work, which is easily done through the U.S. Copyright Office (for contact information for the USCO and other resources, see the sidebar "Getting in Touch").

To register a song, file a Form PA; to register the recording of a song, file a Form SR. If you wrote the song and recorded it, you may file for both copyrights on a Form SR, without filing a Form PA as well. The fee for copyright registration is $30, but you may register more than one song at a time by calling it a "collective work." You can download all the forms from the Copyright Office Web site.

You will need to join a performing-rights society, such as ASCAP, BMI, or SESAC. Once you have some copyrighted tunes under your belt, you are eligible for membership in any of those organizations, but you can join only one. Performing-rights societies' main function is to collect and distribute performance royalties. Those societies also offer excellent resources to members, including regional showcases and educational workshops.

WHO NEEDS IT?

Now the big question: why would any songwriter give away part of his or her most valuable commodity? For many artists, the trade-offs are many and varied. When you sign with a publisher, you gain access to numerous entertainment industry contacts. In short, publishers are responsible for selling the music they represent. They also monitor royalty payments and deal with other legal issues surrounding the use of the songs in their catalogs.

You do. For musicians who are primarily songwriters and desire to have their work recorded by well-known artists, a good publisher is a must. Trying to peddle songs directly to producers is pretty much an exercise in futility unless you're a well-known writer. Many of the biggest names in pop, country, R&B, and adult contemporary music do not

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write their material, which means many record executives and producers spend a lot of time trying to match their artists with the right songs. They often turn to publishers for help in finding them.

If you write and perform your music, you may not be that interested in having other artists cover your songs. You could, however, want much wider exposure for your music, in which case you should consider working with a publisher. Those pros have daily contact with music supervisors for film and television, game producers, and advertising executives. Like the record industry, those areas of the entertainment business are hungry for music. Often, they use the artists' performances of the songs, which means money and exposure for you and your songs.

Even if you don't have lofty goals of worldwide fame, a publisher can be a good partner in your music career. If you don't care to learn many of the legal and business aspects of selling your music, consider teaming with a publisher—let him or her handle the business so that you can focus on your craft.

You don't. If you are mainly interested in writing, recording, producing, and selling your music yourself, you don't have to use a publisher. You may opt to do so for the reasons noted previously; otherwise you can handle it yourself. The positive side of being your own publisher is that you get to keep the songwriter share and the publisher share of the royalties. The downside is that 100 percent of the royalties you earn might not add up to what 50 percent of the royalties for a better-selling song would have earned if your music had reached a wider audience.

You might relish the business end of making and selling music, in which case acting as your own publisher could be a good experience. It is a business, though, so you'll need to approach it as such.

First, name your publishing company and fill out a fictitious business name statement or a DBA ("doing business

as") form with your county. Then educate yourself about the fine points of publishing. For example, how will you collect your royalties? The mechanical licenses for your songs, which make mechanical royalties possible, can be issued through the Harry Fox Agency. In addition to issuing mechanical licenses, Harry Fox also collects and distributes the mechanical royalties and conducts audits to be sure the proper amounts have been paid. The agency takes about 4.5 percent of what it collects.

You also need to register as a music publisher with your performing-rights organization in addition to registering as a writer. By doing so, you can be sure that you get both the artist share and the publisher share of the performance-rights royalties.

Finally, if you're interested in selling your songs, industry executives and music supervisors need to hear them. As complicated as publishing's legal details can be, the song peddling is the hard part. It's like trying to get a record deal on your own: you need to network with industry executives and get that all-important word-of-mouth recommendation that will open doors for you and your music. That is tough to accomplish if you're an unknown.

Open options. Now that you've read the pros and cons of using a third-party publisher versus the DIY approach, here's another option: be your own publisher when your catalog is relatively small, and then sign with a publishing house later in your career. You can opt to sign with a publisher for your entire back catalog (if you own the publishing rights) and future songs, for only the back catalog, or for just the songs written from the point of the publishing agreement forward. Many publishers also offer single-song deals. If you have a composition that is appropriate for certain projects, they will take the publishing rights for just that song.

FINDING ONE

Finding and signing with a publisher can be as tough a task as getting a record deal. You need to produce quality demos of your songs and put together a press kit. Most important, you need a solid vision of what you want to accomplish by signing with a publisher. If you want to be a songwriter for the stars, you should have an idea of what type of artists would be perfect to sing your songs. If you want to write for motion pictures, you should know what type of scene or setting would be a good match for each of your songs.

In short, a publisher will not give you the time of day if your only goal is to get whomever in whatever part of the industry to hear your music and make you rich. If you don't understand your music and your goals, you can't expect anyone else to, either. (See the sidebar "Further Reading and Research" for a list of books that includes guides to domestic and international publishers and several volumes that offer in-depth publishing advice.)

FURTHER READING AND RESEARCH

Whether you act as your own music publisher or sign with a professional, you need to gain some knowledge of how the business works. Hundreds of texts and directories are available on the subject of music publishing. Here are a few books and directories that offer solid information and helpful leads.

The 2001 Recording Industry Sourcebook, 12th ed. (Artistpro.com, LLC, 2001)
The Craft and Business of Songwriting, by John Braheny (Writers Digest, 1995)
The Dictionary of Music Business Terms, by Tim Whitsett (MixBooks, 1999)
How to Have Your Hit Song Published, by Jay Warner (Hal Leonard, 1988)
The Music Business (Explained in Plain English), 2nd ed., by David Naggar (DaJé, 2000)
The Music Publisher Registry, edited by Ritch Esra (The Music Business Registry, 2001)
Music Publishing: a Songwriter's Guide, by Randy Poe (Writers Digest, 1997)
Music Publishing—the Real Road to Music Business Success, 5th ed., by Tim Whitsett (Artistpro.com, LLC, 2000)

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The hunt for a publisher could be tough. With so many music publishers in the business, how do you know where to start? The big players such as Warner-Chappell, BMG, MCA, Sony, and the like are alluring. Those companies are the movers and shakers in the industry, with millions of songs in their catalogs. On the positive side, they do deals daily with major players looking for music. The negative side is that they represent so many songs that your music—should you actually strike a deal with one of those powerhouses—could languish unnoticed among millions of tunes. Small publishers offer much more personal attention to their writers, but some might not have the reputation and contacts necessary to break you into the big time. (For an interesting discussion of how publishers, big and small, approach their businesses, see "Working Musician: Publishers' Roundtable" in the August 1997 issue.)

Research is key in deciding which publishers to send your songs to. Call the publishers and any other music-industry contacts you have and ask which companies specialize in the kind of deal you want. Find out how many songs are in their catalogs and about their track record for song placement. Then narrow the field to five to ten publishers; any more than that could be difficult for you to keep track of as you do your follow-up work. (You can always approach more publishers if some of them from your initial list don't pan out.)

Get the name of the person to give your material to and send him or her a personalized letter, a few of your songs, and a press kit. In your letter, state your goal of getting a publishing deal, show that you're familiar with that publishing house's specialty, and explain why you are interested in working with that publisher. Enclosing a prestamped, self-addressed response card is a good idea—it's an easy way for the publisher to let you know the package was received.

About two weeks after you send the package, call to make sure the right person received it and ask whether he or she has had the opportunity to listen to your work. You might have to call a few times, but don't pester anyone. Polite persistence pays off far more than aggressive rudeness.

DECISIONS, DECISIONS

Songwriters have many decisions to make before even getting started with publishing. Analyze your music, your goals, and how much work you are willing to do and are capable of taking on. Would you appreciate the satisfaction of the DIY approach, or would you prefer the partnership and exposure that come from working with a publisher? It's your music and your decision.

Mary Cosola *is a contributing editor for* **EM.**
Thanks to Michael A. Aczon for his guidance with this article.

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REVIEWS

AKAI

DPS16

An easy-to-use digital studio with a powerful punch.

By Allan Metts

With so many fresh, feature-laden alternatives vying for your attention, all-in-one portable digital studios continue to grow more numerous and powerful. Akai is no stranger to the digital-audio world and certainly familiar in the recording world; Akai tape decks have been available for decades.

For the DPS16, Akai combined powerful recording and mixing capabilities in an easy-to-use package. The DPS16 offers 16 playback tracks with as many as 10 simultaneous uncompressed recording tracks. It features a robust 26-channel mixer, loads of effects, extensive editing capabilities, support for 24-bit and 96 kHz operation, and the ability to gain access to external SCSI drives (including CD-R and CD-RW drives). My review unit had a 10 GB internal hard drive, but models now ship with a 20 GB drive.

The DPS16 sports an attractive and professional-looking exterior, with lots of dedicated buttons, faders, and knobs (see Fig. 1). Measuring roughly 20 inches wide by 14 inches deep, the DPS16 packs a lot of functionality into

FIG. 1: The DPS16 has 16 nonmotorized channel faders and one main fader. Dedicated buttons and knobs are supplemented by soft controls whose functions appear in the tilt-up display. Two of the eight analog inputs accept XLR or ¼-inch plugs and offer switchable phantom power.

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its small footprint. A 6-inch diagonal, 320-by-240-pixel LCD offers plenty of graphic information, and a contrast control makes it easy to see. The LCD tilts to further improve visibility.

Six analog inputs have ¼-inch TRS jacks, and two more have nifty combo jacks that accept XLR or ¼-inch TRS plugs. On the back panel, an RCA jack provides stereo S/PDIF input. Master stereo outputs are available on analog and S/PDIF RCA jacks, and there are four aux sends on unbalanced ¼-inch TS jacks (see **Fig. 2**).

A separate monitor output routes to dedicated RCA jacks and to the ¼-inch stereo headphone jack, each with its own level control—a nice touch. The monitor signal is usually identical to the master output, but as expected, soloed channels are heard only in the monitor signal. The master signal always contains the complete mix.

Rounding out the DPS16's back panel is a 50-pin, half-pitch SCSI connector; a footswitch jack; and two MIDI ports. You can configure one of those for MIDI Out or MIDI Thru.

FIG. 2: The DPS16's uncluttered back panel provides stereo main outputs, stereo monitor outputs, four aux sends, S/PDIF I/O, two MIDI ports, and a SCSI connector.

FIT AND TRIM

Each analog input has a trim control and overload indicator. The overload indicator makes level setting effortless. Just crank up the knob until the LED starts flashing and then back it off a bit. Inputs 1 and 2—the ones with the XLR connectors—have switchable phantom powering, and Input 8 has a high-impedance switch for use with electric guitars.

Sixteen fader groups on the DPS16 correspond to the 16 available recording tracks. Each fader group contains a nonmotorized fader, a pan control, and Track Select and Record Select buttons. You use the Track Select buttons often, primarily when you change settings and route signals. The Record Select buttons let you arm tracks for recording; they glow red when recording is enabled.

Input Select buttons for the ten input channels are above the first ten fader groups. Those buttons are used in much the same way as the Track Select buttons. Their positioning gives the impression that each input records to the corresponding track (Input 1 to Track 1, for example). That is by default normally true, but you can configure the Input-to-Track mappings.

To the right of the fader groups are the Master fader, the Master Select button, and a host of other buttons that give access to the DPS16's features. There are also transport controls, cursor controls, and a Jog/Shuttle wheel. The LCD panel has its own set of soft buttons for access to its settings. In addition, six soft knobs called Q-Link knobs provide quick adjustment of the settings that appear onscreen.

If you get the feeling that the DPS16 has plenty of buttons to press and knobs to turn, you're right. That's what I like about this thing: when I need to gain access to a setting, I do it quickly and intuitively, and I never have to dig through pages of nested menus.

ON THE BIG SCREEN

Each DPS16 screen is accessible by its own dedicated button. You spend most of your time on the Main screen, which features peak and level meters for every track and every input (see **Fig. 3**). Left and right master meters are also furnished. The current location is displayed in SMPTE time code and bar:beat:tick formats, and the recorded audio material is presented in piano-roll format (horizontal bars for all 16 tracks). There are also indicators for disk activity, the available recording time, and the current Project. A Project contains a recording's audio and parameter settings.

Along the bottom of the Main screen are commonly used functions, which map to the six soft buttons below the display. There you can enable punch-ins and looping, modify sync and time-display settings, zoom the track view in and out, set the metronome, and change the meter-display characteristics (pre- versus postfader and master versus monitor signals).

The two remaining soft buttons determine the functions of the six Q-Link knobs. The screen's far right shows six virtual knobs in the same arrangement as the Q-Link knobs. Labels and values for the virtual knobs are displayed, so it's easy to see what you're doing when you grab a knob.

The Q-Link knobs control EQ and effects parameters. Four auxiliary and effects sends and a 3-band parametric EQ are available to each track and input. To change a setting, bring up the appropriate knobs page using the soft buttons, press a Track Select or Input Select button to indicate where you want to make the change, and then turn the knob. That scheme isn't as easy as having dedicated knobs for every track and input, but it's close.

MIX 'N' MATCH

Another commonly used screen is the Mixer. Like the Main screen, the Mixer features peak and level meters and position indicators. The same virtual knobs appear here too, but the rest of the screen provides access to the settings for the tracks and inputs. (The functionality of the Mixer is the same whether you're dealing with a track or an input, so I'll use the more general term *channel*.)

Using the soft buttons in the Mixer screen, you can select two views of the channel settings. One view lets you see a

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selected parameter's value for all channels, and the other view shows a given channel's settings. You can change values using the Track Select or Input Select buttons, the cursor controls, and the Jog/Shuttle wheel. Like the channel strip in an analog mixer, each channel has settings for level, pan, and EQ, and four controls that determine the amount of signal routed to the four sends. The screen provides control for some of the same parameters as those controlled by the Q-Link knobs and track faders. No matter how you change a setting, the correct value is displayed onscreen.

When you change EQ settings for a given channel, the DPS16 shows a frequency graph of the adjustments you make. In this case, a picture is definitely worth a thousand words. A graph of high- and low-EQ shelving—and the width, frequency, and amount of mid-level EQ—is much more informative than a graph showing only Hertz and decibel values.

The Mixer screen has other settings as well. You can disable a channel or send its signal to the ping-pong bus. You can treat the aux sends as two stereo or four mono signals, and you can configure each send as prefader, postfader, or insert. A send can be prefader in one channel and postfader in another, but if you configure a send as an insert, that send is unavailable in the other channels.

PATCHWORK

The DPS16 includes comprehensive signal-routing capabilities that you can configure with the Quick Patch screen (see **Fig. 4**). In this screen you can make any of the ten inputs available for recording or send them directly to the mixer when you play live or mix down live signals (such as sequenced MIDI instruments) in tandem with previously recorded tracks.

The Quick Patch screen also lets you take advantage of the ping-pong bus, a stereo audio bus separate from the master and monitor signals. Use the ping-pong bus to mix down several channels to one pair of tracks—including effects, if you like.

In the Quick Patch screen, inputs that

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FIRST WE LISTEN

route directly to the mixer are connected with thick black lines. The remaining inputs, send signals, or the ping-pong bus can be patched to any track for recording. Making a patch is as simple as pressing a Record Select button while holding down a Track Select button. When you do that, the Quick Patch screen shows a connection between the source you selected and the track you patched to. The Quick Patch screen is intuitive and easy to use.

You can patch outputs as well. Each of the ten outputs—left and right for the master, monitor, and digital outputs, and aux-send outputs 1 through 4—can carry 1 of 26 signals. You can send the right master signal to the left master output or use the four aux sends as dedicated outputs for four selected tracks. The digital output always carries the same signal as the analog master outputs; otherwise, you can completely reconfigure the output routing, a capability that provides a lot of flexibility.

A NEW RECORD

I have described numerous buttons and screens, but you're still wondering how to record, right? Recording is easy, but first you should create a new Project. That establishes your sample rate and bit resolution and lets you associate a name with the audio you're about to record. Assigning a name to your Project lets you recall it intact after you move on to other Projects.

A quick word regarding sample rate and bit resolution: the higher they are, the less audio you can record and play back simultaneously. At 16 bits and as high as 48 kHz, you can have ten recording tracks and 16 playback tracks, but at 24 bits and 96 kHz, you only get six recording and playback tracks. You also get fewer sends and fewer channels with EQ at the higher audio rates.

When you're ready to record, set your levels using the Input Trim controls and overload indicators, and then make sure you're patched up correctly in the Quick Patch screen. Then press the Record Select buttons to arm the tracks you want to record, hold down the Record button, and press Play.

If you're not satisfied with a take, you have 250 virtual tracks at your disposal, and each can have a unique name. You can store successive takes as virtual tracks and assign only the best to physical tracks for playback. You also have 250 Undo levels, but to keep things simple and to conserve disk space, I set the maximum Undo level at 1.

There are comprehensive punch-in capabilities as well. You can start and stop recording during playback (for manual punch-ins) using the transport controls or any standard footswitch. At the punch point, the signal you hear switches from the track source to the live input, as expected. Automatic punch-ins and looping are supported also, and rehearsal mode lets you practice your punches without committing anything to disk.

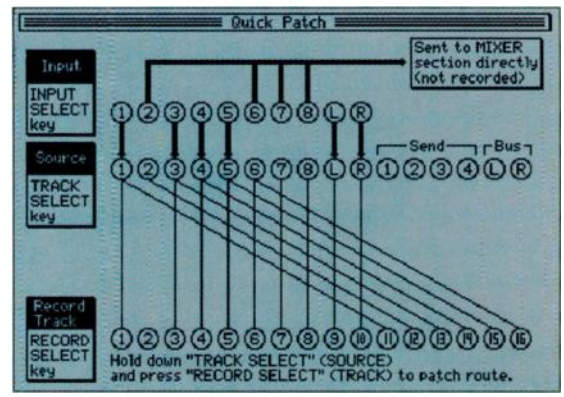
A typical recording session usually requires quick moves from one part of a recording to another. The DPS16 provides 100 Locate points with titles, 26 "quick" Locate points, and a few special locations used in editing and punch-ins.

To mark a spot in your recording, you can use the Record Select buttons to enter time directly in SMPTE time code or in bar:beat:tick format, or you can enter Locate points on the fly. (Numbers are silk-screened beneath the Record Select buttons for entering data.) You can also use the Jog/Shuttle wheel to move in big or small steps. Furthermore, the Play, Fast Forward, and Rewind buttons let you get to points within your recording.

To be precise with your Locate-point placement, open the Waveform screen. There you can zoom in on your audio data and move through it sample by sample. When the Waveform screen is open, the Jog/Shuttle wheel serves as a scrubbing tool to help you find the exact point you're seeking.

Once you find the right spot, mark it with the Memory button. Then you can name it and add it to a list of locations or store it with an Input Select or Track Select button. You can return to marked locations using the Goto button. Locations can be specified during playback, but the process requires two button presses. When you need to mark many points in a hurry, a single button-press option would be easier.

Two special points labeled In and Out are used for punch-ins, looping, and editing. Those points have their own dedicated buttons, and they define the range that will be edited, looped, or recorded over. I especially like the three



COURTESY AKAI

FIG. 4: You can route any input to any track in the Quick Patch screen. You can also send inputs directly to the mixer for live performance or for mixdown of sequenced MIDI tracks.

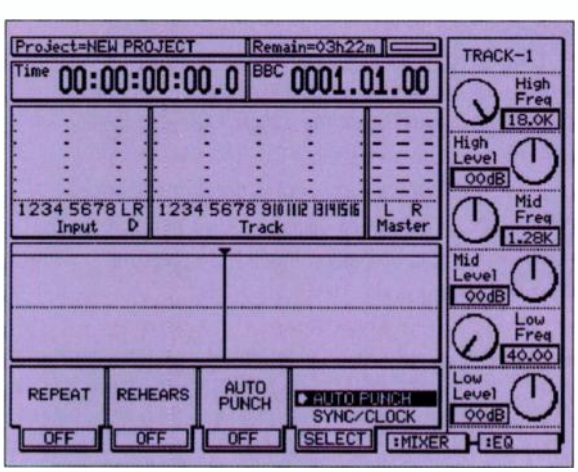


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FIG. 3: The DPS16 Main screen provides peak and level meters for each track and input, a view of your recorded tracks, and plenty of additional indicators.

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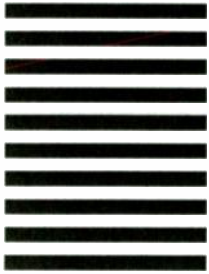

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playback buttons that work with the In and Out points. One plays the time range defined by the In and Out points, and one starts playback a few seconds ahead of the In point and stops when it gets there. The third button plays from one to ten seconds, beginning with the Out point, to help you establish the editing or recording range you want. Using those three buttons, I found it easy to get my punch and editing points just right.

THE EDITING SUITE

The DPS16 has plenty of editing capabilities, which you invoke from an Edit screen. You can cut, copy, insert, overwrite, delete, stretch, and move audio as needed. Multiple tracks can be processed simultaneously. You can copy or move audio from track to track by using buttons to draw connections between source and destination tracks. The DPS16 covers all my basic editing needs.

The time-stretching algorithms are particularly impressive. The stretched or compressed audio isn't always free of artifacts, but as long as I don't try to stretch things too far, I usually obtain satisfactory results. Several stretching presets are offered, with names that convey the intended source material, such as Male Vox A and Lfreq Rhythm B.

You can also use the DPS16's real-time effects processing to alter and enhance your tracks as you record or afterward. You can assign any of 52 effects to one of the four sends. Only one effect can be used in a send at a time, but many channels can use that send if none of them configure it as an insert effect. You can connect your own effects to the DPS16 using the four aux-send jacks. There are no aux returns, however, so you have to use the channel inputs to route an externally processed signal back to the mix.

Each effect type has a dedicated page with the effect's diagram and editable parameters. The variety of effects offers a huge array of parameters, so the explanatory diagrams come in handy. For each effect, the settings you're most likely to modify are made available using the Q-Link knobs, letting you easily change values from the Main and Mixer screens.

The DPS16 features most effect types found in a modern multi-effects unit: reverbs, choruses, flangers, phasers, and delays are all present. A couple of surprises are included as well, such as a rotary-speaker simulator with MIDI speed control and real-time vocal-pitch correction.

Overall, the effects' quality is excellent. The reverbs are smooth, and the choruses are silky. The distortion effect is better than I've heard in other all-in-one studios but not as good as a dedicated guitar processor. The pitch corrector is fun, though it doesn't perform miracles with my lousy singing voice.

SAVE THAT SCENE

The mixer and effects settings can be saved to one of 16 Mixer Scenes, which you can recall with a few keystrokes. Mixer Scenes are useful for putting the mixer into a known state before you start working on a particular part of your Project, but you probably won't find them useful for quick mixer changes during mixdown. For those, save mixer snapshots and controller movements to an external MIDI sequencer or use Akai's *MESA II* software.

Although the DPS16 supports snapshot and real-time transmission of its mixer controls through MIDI, not all the mixer and effects settings can be sent directly from the unit. In fact, you're limited to only level and pan controls for each mixer channel.

If you want to automate other settings, such as EQ or send levels, and you have a Windows-based PC, you're in luck. Akai offers *MESA II* as a free download for its samplers and recorders. *MESA II* provides virtual channel strips for all 26 mixer channels. It's quite helpful to see all the DPS16 mixer controls on one screen.

MESA II also provides a virtual transport control, a beat-map editor, MIDI synchronization support, and access to 20 mixer snapshots. You can record level and pan changes from the DPS16. Using your mouse, you can draw changes to those events and other settings, such as EQ parameters and send levels. You

DPS16 Specifications

Recording Media	internal IDE hard disk; external SCSI hard disk
Physical Tracks	16
Virtual Tracks	250
Simultaneous Recording Tracks	6–10
Simultaneous Playback Tracks	6–16
Analog Inputs	(2) XLR/¼"-TRS combo; (6) ¼" TRS
Analog Outputs	(2) RCA main; (2) RCA monitor; (4) ¼" unbalanced aux sends; (1) ¼" stereo headphone
Digital Audio I/O	S/PDIF coaxial
Additional Connections	50-pin SCSI; MIDI In, Out/Thru; ¼" footswitch
A/D and D/A Converters	24-bit, 128× oversampling
Sampling Rates	32, 44.1, 48, 96 kHz
Sampling Resolution	16-bit, 24-bit
Synchronization	sends/receives MTC; sends MIDI Clock and Song Position Pointer
Effects	52 types: reverb, delay, chorus, flanger, phaser, pitch shift, pitch correction, compressor/limiter, distortion, wah, pan, rotary speaker, noise gate, expander, enhancer
Main Display	320 × 240-pixel, backlit LCD
Dimensions	20.28" (W) × 5.02" (H) × 14.11" (D) (with LCD tilted down)
Weight	14.11 lbs. (without hard disk)

can also use the mouse to revise previously recorded automation data; however, you can't create real-time changes to the effects parameters.

Even without *MESA II*, the DPS16 provides plenty of support for MIDI synchronization. It can act as a master or slave to MIDI Time Code, and it transmits (but doesn't receive) MIDI Clock and Song Position Pointer messages. You can create beat and tempo maps so that the DPS16's bars and beats line up with your sequencer's. (You'd probably create those maps anyway, especially if you use the metronome.) There's even a variable pitch control.

I'LL TAKE IT

The DPS16 is a machine packed with features. Additional capabilities include backup to external devices and the ability to burn CD-Rs. Unfortunately, because my CD recorder is not MIDI Machine Control compliant, I was unable to test the CD burning feature.

The audio quality is as good as or better than similar devices I've heard. I conducted an unscientific A/B comparison between the DPS16 mic preamps and others I own, and the DPS16 held up well; I heard crystal-clear highs and solid lows.

Because the DPS16's operating system can only be upgraded with SCSI, you need a CD-ROM or another portable SCSI drive. The unit ships with a comprehensive Operator's Guide, and Akai offers plenty of Web support, with software updates and documentation. You almost don't need documentation or support with this device. The user interface is incredibly intuitive, and helpful hints are often presented right on the LCD.

I can't find much I don't like about the DPS16: it's portable, flexible, easy to use, and it sounds great. If you're in the market for an all-in-one portable digital studio, check it out.

Allan Metts *is a musician, software and systems designer, and consultant in Atlanta, Georgia.*

PRODUCT SUMMARY

Akai
DPS16
portable digital studio
$2,795

FEATURES	4.5
EASE OF USE	4.5
AUDIO QUALITY	4.5
VALUE	3.5

RATING PRODUCTS FROM 1 TO 5

PROS: Intuitive user interface. Excellent signal routing. Capable recording and editing tools. Loads of effects. Downloadable front-end software (PC only).

CONS: Not enough inputs. Some effects are limited. Effects parameters can't be automated. Mixer scenes and marker placement aren't efficient for quick use.

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Akai Musical Instrument Corporation
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BITHEADZ

PHRAZER 1.0.1 (MAC)

Mac users now have a powerful new tool for getting loopy.

By Jeff Burger

Although loop-based music production has made quite a splash during the past few years, most of the serious action has been on the PC. That is due in large part to Sonic Foundry's popular *Acid* program. Fortunately for non-Windows musicians, BitHeadz has answered the demand on the Mac side of the aisle with the introduction of *Phrazer*—a standalone loop-based arranger that supports all the popular Macintosh audio I/O options, including ASIO, Sound Manager, DirectIO, DirectConnect, ReWire, and MAS.

Like *Acid*, *Phrazer* addresses a fundamental limitation of traditional audio editors: changing the tempo of a sample (such as a drum loop) also changes its pitch. Loop-based programs solve this dilemma by slicing samples into short segments (to isolate rhythmic elements) and triggering the individual slices according to a specified tempo. *Phrazer* doesn't perform surgery on the files. Instead, it adds nondestructive split points (markers) to the audio files and applies them during playback, matching different tempos as needed. The success of tempo changes created through this process has much to do with the quality and nature of the samples: slowing things down may leave gaps of silence between beats; whereas faster tempos sometimes clip the ends off samples.

The number of simultaneous samples that *Phrazer* can play back depends on your computer. I was able to play 15 factory loops along with a few effects using a G3/300 MHz. *Phrazer*'s preferred method of playback, which delivers the best performance, involves preloading the loops into RAM; the program can also stream tracks directly from the hard drive, which could be important if you have limited RAM or if you're working with exceptionally large samples. *Phrazer* requires at least 64 MB of RAM to get started, but more is definitely recommended.

Minimum System Requirements

Phrazer

Power Mac G3/300 (optimized for G4); 64 MB RAM; Mac OS 8.6; 500 MB hard-disk space (for full installation and included library of loops)

FIG. 1: *Phrazer*'s track display appears to the right of its customizable set of controls; a movable divider lets you reveal more of the display. The sample editor at the bottom of the window lets you edit loops and adjust split points.

FASHION PHRAZE

Phrazer's versatile design lets you use it in a number of ways. For example, you can set individual keys on a MIDI or computer keyboard to gate (mute) tracks in real time, which offers some interesting possibilities for live performance or studio work. Moreover, you can record your performance onto your hard drive as a 16-bit stereo AIFF file. You can also render a noninteractive set of tracks to disk at a speed faster than real time. Support for technologies such as ReWire and DirectConnect lets *Phrazer* interact with a sequencer through external sync. (In theory, you can use IAC to route tracks from *Phrazer* into individual channels in a program such as *Pro Tools*, but the process can be rather convoluted.) As of this writing, *Phrazer* does not properly receive external sync through FreeMIDI 1.45; versions 1.43 and 1.44 still work fine. (BitHeadz and Mark of the Unicorn are working on a solution.)

Phrazer imports WAV, AIFF, Sound Designer II, and CD-audio samples. It also reads ACD files from the many available *Acid* libraries, although the *Acid* 2.0 format is causing BitHeadz some problems, which the company is working to resolve. *Phrazer* cannot read Steinberg's *ReCycle* (REX) files directly, but you can save the files in Sound Designer II format from within *Recycle* and then import them into *Phrazer*. You can also easily convert BitHeadz *Unity DS-1* files into a usable format, and you can record your samples from within *Phrazer* as well.

The handy Files window provides a

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FIG. 2: *Phrazer* includes a simple Mixer window with faders, meters, and pan controls.

hierarchical view of your mounted volumes and the compatible files they contain without having to go to the Finder or use the Open command. Clicking on a file name auditions it; double-clicking on it places it on a track. However, the Files window does not automatically update to show new samples created with *Phrazer*'s sample editor; you must click on the Finder and then return to the program before newcomers appear.

Phrazer's main window accommodates a stack of tracks, and each track can contain multiple nonoverlapping samples (see Fig. 1). The track display, however, offers only a rudimentary set of features. For example, a snap-to-grid function ensures the tracks' visual alignment, but no metronome is provided to help you align beats aurally. In addition, you can't simply drag and drop samples between tracks; you must cut and paste them. The track display doesn't scroll automatically to let you drag samples within a track to hidden parts of the timeline.

The main window has a pop-up list that lets you specify which controls appear next to the tracks. Available controls are record-enable, play-enable, MIDI and computer-keyboard assignments, track-color selector, volume, pan, two effects sends, and a meter. Many of the controls take up quite a bit of space, and when combined, they consume much of the room that is needed for displaying the track contents. You can, however, drag a separator to hide some or all of the controls. The program provides a separate Mixer window, but it only includes faders, meters, and pan controls (see Fig. 2).

PHRAZE FIXES

The selected sample appears in the sample editor at the bottom of the main window. There *Phrazer* assigns split points based on amplitude changes in the waveform or based on a designated tempo. The default amplitude-based settings for low threshold, minimum threshold, and minimum time generally get you in the ballpark, but often the split points still require tweaking. The tempo option is a much better bet if you know your loop's tempo.

Other editing features appear in a pop-up menu with which you can specify loop points in several ways. You can select an area in the waveform and choose Set Loop Selection. You can then engage looped playback and make refinements by dragging the loop markers while viewing them at various zoom levels. Alternatively, the Loop dialog box allows you to adjust the loop points with nudge arrows or by entering numbers.

A Munge pop-up menu provides several options for assigning tempo (essential for synchronizing multiple tracks) and key. A variety of other Munge options (destructive edits) include normalize, reverse, parametric and shelf EQ, and flange. To hear the changes you make along with the rest of the song, you must first hit the Save button. Unfortunately, a Save As command is not provided at the sample level, which means you must make backups of your samples before you begin editing—a definite inconvenience.

Phrazer lets you specify how tempo mapping and pitch shifting are applied during track playback for each sample. The default, Split Pitched, uses split points to affect tempo and lets you transpose the music in real time. Split Non-Pitched is the same except that it doesn't allow transposition. Pitch Shift ignores split points but allows transposition. The Raw option doesn't allow manipulation, which is useful for working with samples that have no tempo, such as ambient sounds. The sample editor also determines whether the sample retriggers as a one-shot each time its track is manually triggered from MIDI or computer keys.

PHRAZE EFFECTS

Phrazer incorporates real-time effects at three levels. Two effects sends let you set individual track and master amounts, and two global effects let you process the mix output. You can also insert dedicated effects (along with tempo changes, volume changes, and other events) in tracks. Unlike *Acid*, *Phrazer* allows multiple insert effects to reside in a track, but only one at a time.

The screen area that defaults to displaying the sample editor also toggles to

PRODUCT SUMMARY

BitHeadz
Phrazer 1.0.1 (Mac)
audio loop arranger
$399

FEATURES	3.0
EASE OF USE	3.0
DOCUMENTATION	2.5
VALUE	3.0

RATING PRODUCTS FROM 1 TO 5

PROS: Brings multitrack loop-based composition to the Mac. Allows multiple samples and effects per track. Reads a variety of audio-file formats including ACD. Live triggering of tracks with MIDI and computer keyboard.

CONS: User interface needs work. Poor documentation limits productivity. Arriving at accurate split points for samples can be difficult, especially when synching multiple tracks.

Manufacturer
BitHeadz, Inc.
tel. (888) 870-0070 or (401) 886-7045
e-mail info@bitheadz.com
Web www.bitheadz.com

FIG. 3: *Phrazer*'s effects display replaces the sample editor at the bottom of the window.

show the effects parameters (see **Fig. 3**). The available effects include parametric EQ, shelf EQ, flange, chorus, phaser, filter, dynamic filter, delay, reverb, compressor, distortion, and degrade. The effects' quality is about average.

FIRST PHRAZE

For anyone new to loop-based music production, it's important to realize that *Phrazer* shares the limitations of similar products. You can't simply import loops willy-nilly and expect them to work well together, much less sound good at tempos that deviate dramatically from the original. Moreover, your success in grabbing and slicing your loops depends greatly on skill and attention to detail—especially if you don't know the samples' tempos. As an example, I spent hours in *Phrazer* trying unsuccessfully to get a simple conga slap to loop on the downbeat of the intro drum groove to the B-52's "Love Shack." Well-designed loop libraries, such as BitHeadz's new *Tempo Tantrum*, definitely offer significant advantages.

Phrazer's user interface has several noteworthy shortcomings in addition to those mentioned earlier. For instance, you can't have the transport automatically return to zero after you hit the Stop button; the program also lacks a track-solo function, a sample-revert option, and a keyboard shortcut for auditioning in the sample editor. To make matters even worse, the software's online-only documentation (a PDF file) offers little assistance in helping users become more productive and less frustrated. The program compounds the problem with its lack of online help and pop-up labels. According to BitHeadz, several of those issues are being addressed for the program's next update.

Phrazer definitely feels like a first release—not least because it crashed several times on two different but stable Mac systems. That said, however, *Phrazer* offers a level of multitrack loop-based functionality that has long been missing in the Mac market. It has the potential to change the way you make music, especially if you have a library of high-quality loops.

Jeff Burger *is a songwriter and multimedia producer based in Sedona, Arizona.*

PRESONUS

VXP

This voice processor blurs the line between affordable and high-end.

By Myles Boisen

In the race to capture a share of your gear budget, many manufacturers have recently rolled out flashy, high-powered, channel-strip–style processors oriented toward vocal recording. The most stripped-down hybrid voice processors offer a single microphone preamp, a compressor, and an equalizer; as you move up the scale, the units typically provide better components and more features.

Among budget voice processors that cost less than $1,000, what you typically pay for is a single, proven feature—for example, a good preamp or an acclaimed compressor—with a mixed bag of mediocre extras thrown in. The single-channel PreSonus VXP breaks the budget mold by offering a bountiful selection of high-quality, genuinely useful processing stages, starting with a mic preamp that gives the top contenders a run for their money.

STRING OF TOOLS

With its thick cool blue aluminum front panel and brushed aluminum knobs, the VXP matches the look of the company's MP20 stereo mic preamp. The front panel is divided into six segments according to function and has a big glowing "red eye" push-button power switch on the far right.

From left, first up is the mic preamp section, which provides a continuously variable gain knob with 0 to 60 dB of gain marked in 10 dB steps. Next is the proprietary IDSS control, which ranges from 0 to 100 percent. According to PreSonus, the IDSS control allows for manual adjustment of the drain current on the input FET amplifier, thus increasing even harmonic distortion as IDSS processing is added. That unusual circuit modification is not intended to produce overload distortion associated with guitar amps and fuzz boxes but rather to emulate the thickening effect of even-order harmonic boosting in vacuum tubes.

Just above the gain and IDSS controls is a green eight-segment LED that indicates preamp gain levels at –28, –14, –9, –3, 0, +3, +9, and +18 dB, with a red LED assigned to the +18 dB value. There is no polarity-reverse switch, but 48V phantom power and a 20 dB attenuation pad are selectable from plastic push-button switches that glow green when engaged.

The VXP's compressor portion traces its lineage to PreSonus's popular Blue Max Smart Compressor. But unlike the Blue Max, which offers 15 presets and a manual setting that allows for control over conventional parameters (ratio, attack time, release time, and so on), the VXP's compressor provides only 16 presets: 5 Light, 5 Medium, and 6 Heavy. The presets are maximized for vocal applications, but according to PreSonus, they are useful in other applications as well. The manual doesn't disclose the individual presets' exact parameters, but it does give ratio ranges for the three groupings—Light 1.1:1 to 1.5:1, Medium 1.6:1 to 2:1, and Heavy 2.5:1 to 8:1—as well as general applications.

The compressor's input and output (make-up) gain controls are located on a concentric, dual-pot control. By letting the signal level's adjustment be above or below a preset threshold (determined by the compressor preset), the input-gain knob functions essentially as a threshold control—a rather important point the manual fails to mention. Input gain can vary from –12 dB to +18 dB; make-up gain is adjustable from –20 dB to +20 dB. A backlit In/Out switch (bypass) allows for A/B comparison of compressed and uncompressed signals. An eight-segment meter with green LEDs indicates gain reduction at –1, –2, –3, –4, –6, –9, –12, and –18 dB.

A downward expander—designed for smooth, tapered fades of background noise—is simple but surprisingly effective with just two controls on one concentric shaft. The outer threshold ring adjusts the signal level at which the expander starts to work, covering the entire dynamic range of most audio between its extremes of "off" (–70 dBu) and maximum of +20 dB. PreSonus thoughtfully gave the expander a ratio control, which allows continuous manipulation of gated signals' gain reduction, from a subtle 1:1 setting to the complete muting offered by an infinity:1 ratio. LED metering is provided at four gain-reduction points: –3, –6, –12, and –24 dB.

A dual-function concentric knob also controls the de-esser. The inner shaft selects frequency (continuously variable from 800 Hz to 8 kHz) and is targeted at prominent or overly sibilant *esses* common in many vocal recordings. The de-esser's threshold setting (–40 dB to +20 dB or "off") reduces gain only at the offending frequency and typically passes bass and midrange frequencies untouched so that the de-essed signal loses its sizzle but not its strength. A four-section meter—with green LEDs at –1, –3, –6, and –12 dB—

The PreSonus VXP voice processor provides an excellent Class A mic preamp, a smooth-sounding peak limiter, and a wealth of high-quality signal-processing stages.

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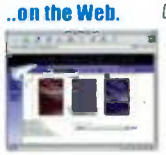

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indicates the module's frequency-specific gain reduction. The de-esser and expander sections do not provide a bypass, but they can be disabled by setting their respective threshold controls to the clearly marked "off" positions (counterclockwise for the expander, clockwise for the de-esser).

The VXP's 4-band equalization is implemented as separate high and low shelving (fixed frequency at 100 Hz and 12 kHz respectively, –12 to +12 dB gain on both) and two concentric, semiparametric midrange controls. The low midrange knob covers three octaves from 90 to 700 Hz, and the high midrange pot spans 450 Hz (marked as .45 kHz) to 5.8 kHz. Bandwidth for each midrange EQ is selectable with a push-button switch, offering Q values of 2.0 or 0.5. A switchable 80 Hz low-cut filter is included, and the EQ section can be bypassed through an In button.

The VXP's master section comprises a peak-limiter threshold control (brickwall type, 0 to +24 dBu or "off"), a master-level control (–70 to +10 dBu), and an output meter identical to the eight-segment LED array in the mic preamp section. No meter indicates gain reduction for the peak limiter.

GOOD CONNECTIONS

The VXP's rear-panel connections are simple: one balanced XLR microphone input, separate ¼-inch balanced TRS send-and-receive insert jacks, and XLR and ¼-inch output jacks for balanced or unbalanced operation, respectively. The insert return can bring a line-level signal into the VXP for processing, and the send jack can also be used when clean output (that is, without channel-strip processing) from the mic preamp is desired. (The manual doesn't mention where the insert return point is in the VXP circuit, but through testing I determined that it comes after the mic preamp and before the compressor and subsequent options.) The line-input level also shows up on the mic preamp input-level meter.

There is no provision for –10 dBV consumer-level output, though the master gain control could easily be used to reduce gain to a level appropriate for –10 input devices. A standard IEC power connector and AC voltage selector are on the rear panel, as is a bay for the optional VXPD2496 digital-output converter and card module ($399).

CARD-CARRYING MEMBER

I tested two VXPs: one with the digital card packaged separately and one with the card already in place. The output card is a stereo converter, meaning that you need to buy only one card to record digitally from a pair of VXPs. Just the same, I wanted to try my hand at installing the VXPD2496 in the stock unit. The module was in place in less than ten minutes, and the only difficulty I had was getting some screws out, thanks to the VXP's solid construction.

The VXPD2496 is truly a marvel of miniaturization. It provides AES/EBU and S/PDIF output jacks, BNC word-clock In and Out, a ¼-inch TRS right-channel analog in, an internal/external sync LED, and separate bit-rate and sampling-frequency selector switches. Green LEDs indicate selected bit rates of 16, 18, 20, and 24, in addition to all common sampling frequencies (44.1, 48, 88.2, and 96 kHz). All that, two PC boards, and a Crystal Semiconductor 5396 converter chip are mounted on

VXP Specifications

Inputs	(1) balanced XLR (mic); (2) balanced/unbalanced ¼" TRS (send and receive)
Outputs	(1) balanced XLR; (1) balanced/unbalanced ¼" TRS
Power Supply	internal
Dimensions	1U × 7" (D)
Weight	8 lbs.
MIC PREAMP	
Frequency Response	10 Hz–50 kHz
THD + Noise (0% IDSS)	<0.003%
THD + Noise (100% IDSS)	>0.5%
Noise Floor	–97.2 dBu (@ +12 dB gain)
Dynamic Range	>115 dB
Headroom	+24 dBu
Maximum Gain	36 dB (+12 to + 48 dB)
Attenuation Pad	20 dB (switchable)
Phantom Power	48V
COMPRESSOR	
Input Attenuation/Gain	–12 to +18 dB
Output Attenuation/Gain	–20 to +20 dB
Presets	16
EXPANDER	
Threshold Range	–70 to +20 dBu
Ratio Range	1:1 to ∞:1
DE-ESSER	
Threshold Range	–40 to +20 dBu
Frequency Range	800 Hz to 8 kHz
EQUALIZER	
Rumble Filter Cutoff Frequency	80 Hz (–6 dB per octave)
Low Shelving	100 Hz (±12 dB
Low Midrange	90 Hz–700 Hz (±12 dB)
Low-Mid Q (Bandwidth)	0.5/2.0 octave (switchable)
High Midrange	450 Hz–5.8 kHz (±12 dB)
High-Mid Q (Bandwidth)	0.5/2.0 octave (switchable)
High Shelving	12 kHz (±12 dB)
MASTER	
Peak Limit Range	0 to +24 dBu
Output Fader	–70 to +10 dB

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a 1½-by-4¾-inch panel that attaches to the VXP chassis with two Phillips screws. The VXPD2496 comes with a three-page manual, which includes installation instructions, brief operating notes, and specifications.

When using two VXPs in tandem, the unit housing the VXPD2496 module is automatically assigned to the left digital channel and the other VXP (or any other preamp, for that matter) must be connected to the TRS right-channel analog input on the converter card. The converter could also be used on a line-level stereo mix by applying the left and right mix-bus outputs to the insert-return jacks of two VXPs.

HOLDS ITS OWN

I put the VXP to work right out of the box for a guitar-tracking session. With the exception of the IDSS circuit, every control and connection point was easy to comprehend without my cracking the manual. The unit weighed more than I expected, partly because of the massive Italian-made torroidal transformer visible through the top vents. On a Fender guitar/Fender amp rig, miked with a Royer R-121 ribbon microphone, the VXP issued a solid, authoritative tone that matched a previous track recorded with the same mic through a Drawmer 1960, one of my favorite guitar preamps.

A female vocalist sounded deliciously airy through a BLUE Bottle microphone (with B7 capsule) paired with the VXP. For her lead vocal and background tracks, I used some limiting and low-shelving EQ to polish the signal going to tape, along with the expander to minimize headphone bleed. The modules were responsive and easy to use. However, I probably dialed in conservative settings because of the EQ section's lack of gain markings and metering for the limiter. In that regard, as with many other preamps I've used, the VXP is more an intuitive rather than an exacting unit. Although I'm happy to rely on my ears when adjusting a knob, having calibrated markings on the VXP's parameters would be nice.

I also compared the VXP to a Neve 1272 preamp on an acoustic-guitar lead miked with an Oktava MK 219 condenser mic. The VXP didn't have the up-front attitude of the vintage Neve, but it certainly held its own. It even added some heft to the tone picked up by the rather bright MK 219. I was consistently impressed by the VXP's ability to stand in for much more expensive preamps during the testing phase. Although I generally avoid dynamic processing and equalization going to tape, most of the VXP's processing options met or surpassed my expectations in terms of professional sound quality, low noise, and absence of coloration. My only gripe is about some of the sections' physical implementation—specifically, the concentric pots required to fit everything onto the VXP's crowded 1U faceplate.

Gain controls for the shelving and bell EQ are detented at their zero settings, but curiously, the compressor and master-gain knobs are not. On the compressor, even with perfect visual alignment of the input and output knob settings at zero, engaging the module on below-threshold signals changed the overall level and introduced switching noise.

In addition, the 16 compression presets are not labeled, which complicates repeatability. If it were my unit, I'd be tempted to mark the 16 positions with a grease pencil, for example, L1, L2, L3, and so on.

With compressor gain at unity, a 0 dB line input produced an average of –1 dB gain reduction in Light programs, –2 to –3 dB in Medium, and –3 to –6 dB in Heavy. Light to Medium compression could be relied on for good results, with no negative impact on a variety of signals. But at –6 dB in the Heavy settings, the VXP compressor started to pump audibly and it imparted a sibilant edge to a full-spectrum music mix.

The relatively simple de-esser and expander functions are useful and perform on par with the best-available modules found in other multi-effects dynamics processors. The de-esser is easy to fine-tune, but the lack of a bypass

The VXP's rear panel includes separate ¼-inch TRS send-and-receive jacks and a slot for the optional VXPD2496 converter card.

switch makes A/B comparisons awkward. Considerable finesse is required to twist the outer ring, which controls threshold, to the off position without accidentally moving the inner knob, which selects the de-ess frequency. Furthermore, that dial's only intermediate numerical value is a zero marking at about two o'clock, which again complicates repeatability. Similarly, the expander controls have no intermediate marks except for a 2:1 ratio designation.

FULL STEAM AHEAD

The VXP's 4-band equalizer—a first on any PreSonus product—is also quite effective and remarkably lavish, especially considering the unit's price. Although there was no special magic in the high-shelf range, the low-shelving control let me make beefy bass boosts without adding unwanted flab to a mix. In addition, the high Q option provided enough surgical precision to reveal previously unheard mixing and mastering flaws in some of my older recordings. Thankfully, all EQ knobs are zero-detented and there are helpful frequency markings around the perimeter of the split-shaft bell EQ controls.

Lack of gain-reduction metering aside, the PreSonus limiter sounds amazingly good under all conditions. During sessions it was always subtle and completely transparent, putting a soft touch on peaks without any coloration or grit. Even when I cranked it to the extreme, I noticed no distortion in a full-music mix. Judging by the VXP's output meter and metering on the DAT, the limiter's gain ceiling is not the inflexible type found on most brickwall-type limiters. What the VXP lacks in control-knob precision, it gains in intelligent designs that sound good.

The VXP's only disappointing feature is the IDSS control. In loudspeaker and session tests, I heard no favorable coloration added by that circuit. At 50 percent, the process is audible, serving to take a little edge off a signal—as would a deep-cut, high-shelving EQ set to a corner frequency of 2 kHz. At 100 percent,

it significantly dulled and attenuated a full-music mix and produced a thick- and muddy-sounding midbass boost that pushed the kick drum forward. Granted, that circuit is probably not intended for stereo-program material; but even on individual tracks, the IDSS produced no audible enhancements.

NOT QUITE CONVERTED

During a loudspeaker test for compatibility and sonic character of the VXPs through the onboard converter card, matching the units' output levels using knob calibrations and the LEDs was easy. Because the preamp gains are continuously adjustable, minor tweaks to the left-right balance were a cinch. For stereo recording to digital media, the VXPs provided dramatic headroom, given that output levels ranging from 0 to +3 dB produced maximum levels on my Tascam DA-30 meters.

I also compared the inexpensive VXPD2496 to the A/D on the Apogee PSX-100, a much higher-end converter. For that test, I recorded music samples to DAT through the VXPD2496, followed by the same samples with identical preamp settings through the PSX-100. With the Apogee converter as the only variable in the signal chain, the level to DAT dropped dramatically. It didn't take long to figure out why—the VXPD2496 was simply calibrated to a much higher reference level than usual. PreSonus explained that the latest revision of its A/D converter card was adjusted to conform to the standard reference level of 0 dB equals –18 dBFS. PreSonus sent me an upgraded VXPD2496, and it performed as promised. But with either VXPD2496 version, the unit's +24 dB of headroom is sufficient to drive analog or digital inputs to maximum levels, regardless of the converter or reference level used.

With levels equalized through the PreSonus and Apogee converters, samples from Steely Dan's "Green Earrings" revealed intriguing differences. Tonally, though the snare drum was edgier through the VXP digital card, the two converters sounded close to identical when recording the same program at maximum levels to 48 kHz/16-bit DAT. The major discrepancy was in the reverb on Donald Fagen's voice. The reverb was lush through the Apogee PSX-100, and it always sustained through the breaks between vocal lines. The same section when heard through the PreSonus converter, however, sounded like a much drier mix. The reverb decay was shorter and didn't bridge the gap between lyrics.

That result suggests potential resolution problems in the onboard VXP converter. The higher bit rates in the VXPD2496 will improve the unit's resolution, but based on my comparison tests, I would be wary of using the PreSonus converter for professional classical recording and other critical stereo-recording applications that re-

PRODUCT SUMMARY

PreSonus
VXP
voice processor
$799

FEATURES	4.0
AUDIO QUALITY	4.5
EASE OF USE	4.0
VALUE	5.0

RATING PRODUCTS FROM 1 TO 5

PROS: Many fully professional features at a budget price. Excellent mic preamp with lots of gain. Ample +24 dB headroom. Very useful compressor, de-esser, expander, and equalization functions. Excellent limiter circuit. Insert allows send and return and line-level input. Single VXPD2496 converter card works for stereo signals.

CONS: Compression presets offer limited options. Some processing sections don't have bypasses for A/B comparison. Not all gain pots are detented. No meter for limiter. No polarity reverse. IDSS effect produces no audible enhancement. VXPD2496 converter card doesn't convey low-resolution details well.

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quire accurate room-sound reproduction and subtle ambient details.

MANUAL OVERDRIVE

The VXP manual, though adequate and well intentioned, unfortunately suffers from omissions and errors. Aside from the lack of specifics about the compression presets and use of the compressor input-gain control, the manual mistakenly groups condenser and ribbon mics, stating that "condenser and some ribbon microphones require external power to preamplify the microphone acoustic pickup. These microphones typically have much higher output than dynamic microphones." Not only do ribbon mics typically produce lower output than ordinary dynamic microphones but they're also actually dynamic mics. Furthermore, phantom power should not be applied to ribbon microphones.

The manual also makes the highly debatable assertion that "the –20 dB pad is almost always necessary when close-miking." I especially took issue with the following statement about the IDSS circuit: "This remarkable effect gives you the sound of a tube without the headache of uneven performance often encountered with vacuum tube devices."

HEADS UP

The PreSonus VXP is a powerful, high-quality voice processor with loads of features yet a surprisingly modest price tag. It provides a Class A mic preamp, an IDSS circuit said to emulate tube distortion, a smart compressor with 16 presets, an expander, a de-esser, a 4-band semiparametric equalizer, and a peak limiter. All the processing stages but one, the IDSS circuit, are well implemented, useful, and highly usable. I was impressed with the VXP's mic preamp, which provides as much or more clean gain than many top-dollar units. At microphone or line level, there is plenty of headroom for analog or digital output (the latter for using the optional VXPD2496 converter card). I was also impressed with both the flexibility and transparency of the VXP's peak limiter. Personal-studio operators and picky professionals should note the VXP's competitive sound quality.

To pack such a dense array of features into the VXP's single-rackspace control surface, PreSonus used concentric dual-function pots in the dynamic and EQ processing sections. The face is still a bit cluttered, meaning you have limited finger space and calibration marks are sometimes sparse and hard to decipher. To fit in all the marks, meters, and switches I'd like to see, the VXP would have to be a bigger beast with a larger price tag. As it is, the drawbacks are minor compared with the wealth of high-performance circuitry PreSonus squeezed into an affordable package.

Myles Boisen *is a guitarist, producer, and composer, and is head recording/mastering engineer at Guerrilla Recording and the Headless Buddha Mastering Lab in Oakland, California. He can be reached through e-mail at mylesaudio@aol.com.*

AARDVARK

DIRECT PRO 24/96 (WIN)

A high-quality, affordable digital-audio system for the personal studio.

By Zack Price

Personal-studio owners often face a dilemma when it comes to buying audio cards for their computers. Less-expensive cards never have enough inputs and outputs, but professional-level cards typically offer more I/O than you need in a small studio setup. Why pay for features you'll never use?

Fortunately, Aardvark has introduced a new system that aptly addresses the personal-studio user's needs and budget. The Direct Pro 24/96 offers plenty of analog inputs and outputs without going overboard. It also includes S/PDIF inputs and outputs, a headphone monitoring jack, and software that turns the Direct Pro 24/96 package into a truly self-contained studio system.

FIG. 1: Aardvark's Direct Pro 24/96 audio card provides S/PDIF jacks on the faceplate and built-in real-time effects. The PCI card attaches to a breakout box with a 6-foot, 25-pin cable.

THE HARDWARE STORE

The Direct Pro 24/96 consists of three parts: the PCI host card, the audio interface box, and the connecting cable. The card's solid construction and resin-encased processor conveys quality and reliability (see **Fig. 1**). The powerful digital signal processor (DSP) provides standard digital-audio processing operations and compression, EQ, reverb, as well as a mixing control-panel application for real-time input monitoring.

The card's faceplate provides S/PDIF input and output jacks on coaxial (RCA) connectors. Surprisingly, the S/PDIF jacks aren't located on the interface box, where they would have been more convenient. Instead, the S/PDIF connections are on the card, where they're easier to daisy-chain when multiple cards (as many as four per system) are installed. The first card acts as the master, running on its internal clock. A slave card, set to external clock, syncs to the master card by connecting the master's S/PDIF out to the first slave card's S/PDIF in. That slave card's S/PDIF out then connects to the S/PDIF in of the next slave card, and so on.

A 25-pin port connects the card to the Direct Pro interface box, which houses the analog I/O and the corresponding 25-pin input connector. The 6-foot connecting cable is a standard straight-through pin-to-pin shielded cable, so it might be possible to replace it with a high-quality cable as long as ten feet.

The Direct Pro interface box has four Neutrik combination jacks on the front panel (see **Fig. 2**). Those inputs can accept XLR mic, as well as ¼-inch balanced (+4 dBu) or unbalanced (–10 dBV), plugs. A phantom-power switch supplies 48V to all four XLR inputs for professional condenser mics and DI boxes that need phantom power. The mic preamps on those four inputs are clean, quiet, and well worth the difference in price between the Direct Pro 24/96 and the less-expensive Direct Pro LX6 (see the sidebar "The Lighter LX6"), which lacks the preamps and XLR connections. The XLR mic inputs on the Direct Pro 24/96 have 60 dB level trim, so you can't plug line-level signals in to them; you must use ¼-inch plugs for that.

The interface box's front panel also provides a headphone jack that is optimized for headphones with 60 to 150 ohms impedance. When simply listening to audio playback from a digital-audio application, the headphone output level is comfortable, but you can easily hear what is going on around you, even with headphones designed to minimize outside noise. That could be bothersome during recording situations. Unfortunately, no front-panel volume control to adjust headphone levels is provided; the software control panel handles that function.

The rear panel features four ¼-inch jacks that provide balanced or unbalanced outputs. Their levels (+4 dBu or –10 dBV) are set in the Direct Pro software. In addition, two RCA jacks serve as auxiliary "consumer level"

PRODUCT SUMMARY

Aardvark
Direct Pro 24/96 (Win)
audio interface
$699

FEATURES	4.5
EASE OF USE	4.0
AUDIO QUALITY	4.5
VALUE	4.5

RATING PRODUCTS FROM 1 TO 5

PROS: Excellent-sounding digital-audio system. Clean mic preamps. Wide variety of drivers. Includes well-designed mixer software. Onboard real-time DSP. Direct monitoring.

CONS: No front-panel volume control for headphones.

Manufacturer
Aardvark
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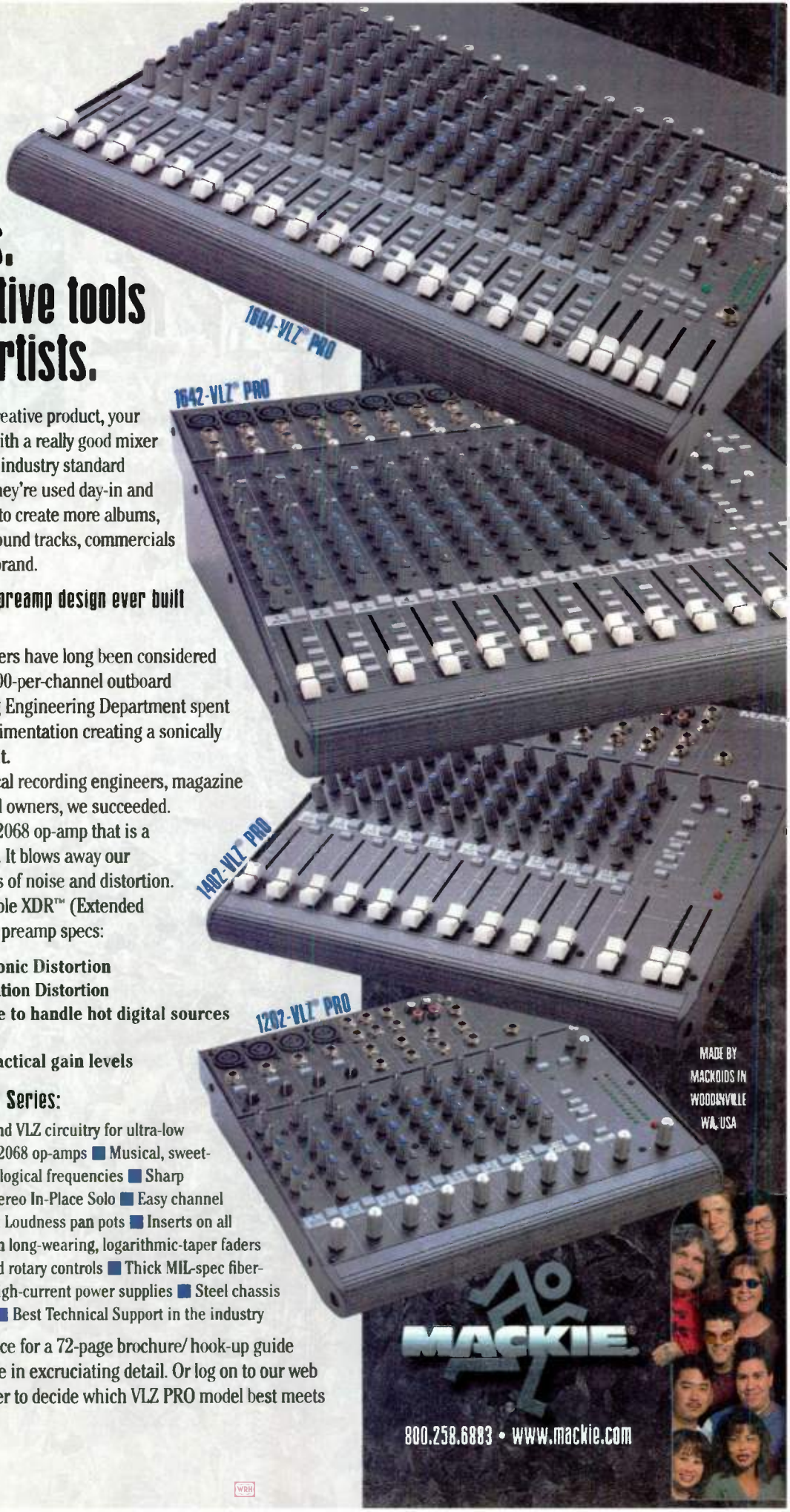

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outputs (–10 dBV). Those outputs can function as the monitor mix for all inputs, outputs, and effects, or as a third pair of outputs (5 and 6), which are handy for surround-sound mixes. A set of MIDI ports on the rear panel lets you connect MIDI instruments or use the interface to produce MIDI Time Code if your digital-audio software supports that capability.

DRIVE TIME

The Direct Pro system comes with MME and DirectX drivers for Windows 95, 98, and ME; ASIO drivers for VST applications; and GSIF drivers for NemeSys *GigaSampler* and *GigaStudio*. Drivers for Windows 2000 and the Mac OS should be available soon. The system is fully compatible with Sound Blaster and most other audio cards. Because of hardware conflicts, however, you can't use the Direct Pro 24/96 with Echo cards or Digidesign's Digi 001 system.

For best performance, adjust the size of the Direct Pro's audio buffers if you plan to play VST instruments in real time using Direct Pro's audio and MIDI capabilities. The default ASIO driver values have too much latency (29 ms); reducing the buffers' sample size brings the latency down to a respectable 12 ms or less. (According to Aardvark, an enhanced ASIO driver with latency as low as 4 ms should be available by press time.)

GigaSampler users will discover that they can play only 16-bit audio, even though the card and the program are quite capable of 24-bit output. That is not a problem with the newer NemeSys *GigaStudio*, however, because the Direct Pro outputs *GigaStudio*'s audio at 24-bit resolution.

CONTROL-PANEL COMFORT

The Direct Pro's construction and audio quality are indeed impressive, and the software that manages the system is equally so. The Direct Pro's Control Panel window looks and acts like a 10×2 hardware mixer, letting you record four channels at once and monitor audio inputs and outputs with zero latency (see **Fig. 3**).

The four input channels on the left correspond to the physical inputs on the audio interface box. Input channels 3 and 4 can be switched between the third and fourth analog inputs and the S/PDIF inputs on the card. The S/PDIF inputs, however, can't be used as two additional inputs along with all four analog inputs.

The six channels in the middle of the mixer are dedicated to playing tracks from a multitrack digital-audio program; each channel can play more than one track. For example, you can take six rhythm tracks playing back from Steinberg's *Cubase VST* and mix them in real time with four live mic or line inputs. The total number of tracks depends on your computer's CPU, available RAM, and hard-drive speed.

FIG. 2: The Direct Pro 24/96 breakout box sports four combination input jacks and a headphone connection on the front panel. The rear panel provides four ¼-inch and two RCA outputs, along with MIDI I/O.

Each of the four input-channel strips provides adjustable mic- and line-level trim settings, as well as a simple but effective compressor with threshold, ratio, attack, and release controls. In the middle of each channel strip, a 3-band EQ offers a high-frequency control centering on 8 kHz, a low-frequency control centering on 220 Hz, and a midrange control with a frequency sweep between 50 Hz and 15 kHz. You can boost or cut all three frequency bands by 12 dB.

The fader section of the four input and six playback channels features several common controls, including Reverb knobs that control the amount of reverb sent to the channel strips. The input strips provide individual faders along with level meters and pan sliders. The stereo playback channels provide left- and right-channel faders and their corresponding level meters. All the input and playback channels include Mute and Solo buttons.

Stereo buttons to the left of inputs 1 and 3 lock the corresponding channels to their neighbors on the right to form stereo pairs. The Pan controls become locked hard left and right, and changes made to either channel in the stereo pair are duplicated in the other channel.

One advantage of using the Direct

Direct Pro 24/96 Specifications

Resolution	24-bit
Sampling Rates	32, 44.1, 48, 96 kHz
Frequency Response	7 Hz–44 kHz, ±0.5 dB @ 96 kHz
Dynamic Range	110 dB (D/A); 100 dB (A/D)
THD+N	0.002% @ 1 kHz
Analog Inputs	(4) combo connectors: XLR mic inputs (60 dB level trim) with phantom power (48V); ¼" unbalanced line inputs at –10 dBV (30 dB level trim); ¼" balanced line inputs at +4 dBu (30 dB level trim)
Analog Outputs	(4) ¼": +4 dBu/–10 dBV; (2) aux RCA: –10 dBV
Digital I/O	24-bit S/PDIF; optional AES/EBU
MIDI I/O	(1) In, (1) Out
Sync I/O	S/PDIF digital clock, MTC
Dimensions (breakout box)	7" (W) × 2.25" (H) × 8.25" (D); 2U rack-mount adapter optional
Weight (breakout box)	4 lbs.

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Pro's real-time effects is that they don't burden the computer's CPU, because the effects are handled by the onboard DSP. Another important advantage is that you can use the built-in effects without recording them; for example, I like to add reverb when monitoring a vocal track or use EQ as a rough guide for determining later EQ settings without actually recording the effects. However, you can engage the Record FX button on any of the four input channels and then record the effects in real time to the corresponding tracks, though you can't use the built-in effects when recording at 96 kHz.

FIG. 3: The Direct Pro's Control Panel window emulates a 10×2 hardware mixer, which lets you control the four physical inputs, the system's onboard effects, and six playback channels from a multitrack audio application.

Preset buttons on the four input channels let you save and recall the current effects settings. However, you aren't limited to recalling the input effects on their original channels; settings that have been used on one channel can be recalled on the other channels too. An overall Preset button, located just below the Master Reverb section, lets you save the current mixer settings, including fader and pan levels, EQ and compressor settings on each input channel, and reverb parameters. The mixer also includes buttons that let you bypass the effects in the input channels.

The Direct Pro 24/96 employs an intuitive drag-and-drop approach to its internal patch bay. Clicking on the Patch

THE LIGHTER LX6

If you don't need the mic preamps that come with the Direct Pro 24/96, you might prefer the Direct Pro LX6. It uses the same card and includes the same DSP capabilities as the Direct Pro 24/96. The analog inputs, however, are line-level only with ¼-inch jacks that accept balanced or unbalanced plugs. The LX6 also provides a volume-control knob for the headphone output (see **Fig. A**).

The mixer software for the LX6 is identical to the Direct Pro 24/96 software except that the LX6 doesn't have mic input level controls. Instead, its software switches levels between +4 dBu and –10 dBV on the input channel strips.

The basic LX6 package includes Samplitude Project for $499; the LX6/*Cakewalk Pro Audio* 9 bundle costs $579.

FIG. A: The Direct Pro LX6 system uses the same PCI card as the 24/96 but lacks mic preamps and XLR inputs. It does, however, have a headphone volume control.

Bay button opens the Patch Bay dialog box. As with hardware patch bays, the Patch Bay dialog box lets you reroute audio signals without constantly plugging and unplugging cables. The Direct Pro patch bay, however, also controls virtual inputs and outputs from your multitrack audio program. For example, you can route analog inputs 1 and 2 to their corresponding analog outputs, or you can connect playback channels 1 and 2 from your digital-audio program to the Direct Pro's analog outputs. To monitor the mix of input and playback channels at the same time, drag the monitor connection to any of the desired outputs. Outputs 5 and 6 are the best choices for monitoring, because they're duplicated at the headphone jack.

You can configure the patch bay in many ways; the manual illustrates a number of common configurations. I have one small complaint, though: to disengage an output, you must connect a cable from the Silence input to the appropriate output. That's inconvenient if you want to erase all connections and start from scratch. As a work-around, set up a Silence preset; then you can easily recall it whenever you want. Fortunately, the patch bay lets you create as many presets as you need.

Minimum System Requirements

Direct Pro 24/96

Pentium/200; 64 MB RAM; Windows 95/98/ME; PCI slot

THE SUM OF THE PARTS

The Direct Pro 24/96, in keeping with Aardvark's reputation, is an excellent-sounding digital-audio system. Its mic preamps are quiet and clean, as are the analog inputs and outputs. The card has a wide variety of drivers covering almost every piece of Windows audio software you could imagine. The forthcoming Windows 2000 and Macintosh drivers will make the Direct Pro even more inclusive.

Furthermore, because the Direct Pro 24/96 uses its own DSP and control panel, you can use the breakout box as a standalone mic preamp and effects processor. You must have the computer turned on and the software control panel running, but the capability can come in quite handy when you're not recording to the computer.

The Direct Pro's mixer program, unlike some software mixers that are bundled with audio cards, is extremely useful. Although it adds a layer of complexity between the Direct Pro 24/96 card and your digital-audio applications, the mixer program's power and flexibility more than make up for any inconvenience. Finally, the Direct Pro's onboard real-time DSP, the direct monitoring capabilities, and the included full version of *Cakewalk Pro Audio* 9 really make this package hard to beat.

Zack Price *occasionally writes under the extremely unimaginative pseudonym of Zack Price.*

EVENTIDE

ORVILLE HARMONIZER

Eventide's latest effects box is a beefed-up, heavy-hitting signal processor.

By Peter Freeman

During the five years since Eventide introduced its popular DSP4000 UltraHarmonizer, processors have become considerably faster and cheaper. The company has capitalized on that trend in a big way with the Orville Harmonizer, its new flagship digital signal processor. The Orville is a fully programmable, multichannel, multipurpose, 24-bit device that shows a clear resemblance to its venerable predecessor, the DSP4000 (see **Fig. 1**). But the Orville goes far beyond the DSP4000's capabilities.

For starters, the Orville incorporates two independent Motorola 56303 signal processors (DSP A and DSP B), each with four virtual inputs and outputs. You can run the two processors in parallel, in series, or in various combinations. In addition, the Orville boasts four analog inputs, four analog outputs, four digital inputs, and four digital outputs (all available simultaneously), and you can route any input to any output (see **Fig. 2**). By combining its highly flexible I/O routing scheme with a staggering number of high-quality presets, Eventide has created a device with mind-boggling potential.

As a previous DSP4000 owner, I often couldn't combine patches because my patches used most or all of the unit's available digital signal processing (DSP) power. With the Orville's significantly greater resources and dual-machine architecture, however, combining patches is easy. (It's possible to combine two preexisting DSP4000 patches and run them on a single Orville processor.)

Not only does the Orville have more DSP power than its predecessor, but it also has much more delay and as much as 174 seconds of sample RAM. That RAM is separate from the 80 seconds of mono delay memory (40 seconds for each processor) that is also included. In fact, DSP A can use the sample time along with its 40-second delay allotment for a total of more than three and a half minutes of delay time.

The Orville supports a wide range of sampling frequencies as well as 16-bit and 24-bit resolution. The 24-bit resolution is hardwired, so if an input signal has only 16-bit resolution, the lower 8 least-significant bits are left blank to be filled by processing. Orville can output a 24-bit signal or dither from 24 to 16 bits. The preset sampling rates supported are 44.1, 48, 88.2, and 96 kHz, though the unit can lock to any incoming sampling rate within that range.

WORKING THE BOX

The Orville continues the now-familiar Eventide interface design, which includes a large backlit LCD, a rotary encoder knob, dedicated and soft function keys, and a numeric keypad. Like the DSP4000, the Orville provides Program, Parameter, and Select buttons and a PC Card RAM/ROM slot for adding or saving new patches. The Orville's expanded I/O capabilities are evidenced by its four LEDs, which are switchable to indicate analog, digital, or internal sources. In addition, the Bypass button functions independently for each processor. Because the Orville's LCD only shows one processor's parameters at a time, a Processor Select A/B button determines which processor's information is visible.

At its most basic level, the Orville is simple to use. You can easily choose a processor, call up a patch, and begin editing with the Parameter button, the four soft keys, and the rotary encoder. That is the same approach that Eventide has used for many years, and besides being familiar to many studio owners, it's efficient and speedy. Depending on the patch you select, there can be a fair number of parameters to tweak, even without delving deep into the editing operations. (Multitap delay programs, for example, may contain individual delay time, level, and filter settings for each of the eight taps.)

The sheer complexity of the Orville's routing capabilities can sometimes be confusing, and the LCD's text-based parameter pages don't make them any clearer. It's too bad the Orville doesn't employ its onboard graphic-display features more extensively.

At first I had a hard time grasping the routing scheme. For example, the Orville flashes a complaining LED if it thinks it needs a digital clock signal from a specific input. I set up a simple series A-to-B configuration using only inputs 1 and 2 of the two processors, but the Orville thought it wasn't getting the appropriate sync.

After a bit of head scratching and a call to Eventide, I figured out that with the Orville, connected routings are

FIG. 1: The Orville Harmonizer's front panel has the familiar Eventide look, including dedicated buttons, soft keys, a numeric keypad, and a large rotary knob for data entry.

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active regardless of signal presence; therefore, you must disconnect unused routings in the routing page. Even if you're only using the digital ins and outs, make sure that the analog I/O is not routed to or from anywhere else on the Orville.

FIG. 2: The Orville's back panel boasts four analog inputs, four analog outputs, four digital inputs, four digital outputs, and remote control and MIDI I/O.

PATCHES TOGETHER

The Orville has far too many effects programs (more than 900) to describe, so I'll highlight a noteworthy few. (For an overview of the Orville's onboard effects, see the sidebar "The Orville's Parade of Presets.")

Eventide has always been best known for its industry-leading pitch-shifting technology, so it's no surprise that the Orville's pitch-shifting capabilities are truly awesome. For example, you can change pitches over an eight-octave range (four octaves up and down) with independent shifters (each with its own delay time) operating simultaneously within an Orville patch. In fact, one of the pitch-shift presets, Twelve Shifts, provides 12 shifters and quad operation. Keep in mind that it uses only one of the Orville's two processors, so loading the Twelve Shifts patch into processors A and B would provide 24 independent

THE ORVILLE'S PARADE OF PRESETS

This partial list of Orville Harmonizer effects categories provides a glimpse into the powerful capabilities of an amazing device. The unit ships with more than 900 presets, and updates should make that number even larger.

Basics: stripped-down starting-point versions of reverb, compression, delay, pitch shift, filtering, EQ, sampling, and delay

Beat Counter: delay programs with an intelligent beat-interpretation mechanism that can extrapolate delay time based on almost any rhythmic input when fine-tuning parameters are set

Delays: parallel, ping-pong, multitap, reverse, and many others

Delays–Effects: more esoteric delay flavors, such as filtered-band delays

Delays–Loops: multitrack RAM-based delay loopers and recorders

Delays–Modulated: flange, chorus, detune, panning, Leslie simulator

Dual Effects: stereo reverb combinations and mixed effects

Dynamics: compressors, duckers, tremolos

Equalizers: graphic EQ, 8-band, 3-band, stereo, quad, and many more

Film–Atmospherics: complex atmospheric programs

Filters: Creamy Vocoder, Harmonic Enhance, Bandpass, EZ Leslie, Cup Mute, and others

Fix Tools: pitch correction for repairing pitch problems on single tracks

Front of House: programs for live sound, such as banks of compressors and instrument-specific reverbs

Inst–Clean: various preamp programs with effects

Inst–Distortion: distortion patches

Inst–Polyfuzz: complex distortion patches with effects

Manglers: patches that degrade input-signal quality, such as bit-depth reduction

Mastering Suite: stereo compressors intended for mastering applications and effects such as Class A Distortion and Tape Flange emulation

MIDI Keyboard: patches that respond in specific ways to MIDI keyboard input, such as harmonization, pitch/delay, and others

MIDI Clock: specific programs that derive timing information from MIDI Clock, such as delays, chorus, panners, and tremolos

Mix Tools: multi-effects toolbox patches intended for mixing situations

Multi–Effects: more effects combinations

Panners: Auto, Circle, 3-D Circle Delay, Fly-by, Gyroscope, Joystik, Quad

Percussion: drum- and percussion-specific effects such as delays and reverbs

Phasers: Static, Random, Sample and Hold, Techno

Post Suite: a unique category in which the Orville is used as a sound source; intended for film post-production—record scratch emulator, whoosh maker, 16 mm–projector simulator

Remix Tools: triggered versions of certain effects, including filtering, flanging, phasing, and panning

Reverb: Halls, Plates, Rooms, Small Rooms, Preverb, Quad, Unusual

Ring Mods: Envelope, Modulating, True ring modulator

Samplers: Timesqueeze, Multi Trigger, Panning, Varispeed, Triggered Reverse, VocalFlyer, Filter Trig

Shifters: normal, Diatonic, Ultra, Unusual

Sound Effects: another synthesis bank—Doorbells, Helicopters, UFOs

Spatialization: 3-D PhaseInverter, Quad, QuadDlyBasedPan

Synthesis: sample-and-hold FM, FM Timbre Factory, Rise or Fall Oscillator

Tap Tempo: delays, tremolos, filter, and other effects with Tap Tempo capabilities

Test Tools: signal generators, scopes, spectrum analyzers

Utilities: tuners, dither, metronome, delay calculator

Vintage Emulation: emulations of well-known pieces of outboard gear, such as the AMS DMX-1580S delay, EMT plate, and reverb

Virtual Pedals: simulations of stompboxes

Vox: vocal-specific effects, including pitch correction, pitch change, delays, and reverbs

Programming: tutorial patches designed to aid in the patch-creation process

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shifters. Add the Orville's vast internal signal-routing flexibility, and you begin to get an idea of its astounding capabilities.

One of the Orville's biggest improvements over its predecessor is the addition of formant shifting. Although that has become a rather trendy feature, it's powerfully implemented in several Orville patches. Ultra Interval, for example, is a corrective shifter with a three-octave range. It lets you specify a separate formant value for each semitone above the three octaves, making it a brilliant corrective tool for vocals. At the opposite end of the applications spectrum is formant manipulation, which can be used as an effect with modulations. Because the Orville's formant parameter can be addressed like any parameter, it is accessible not only by modulation sources within a patch but also by external sources such as footpedals and MIDI controllers.

Since Eventide's H3000, the Harmonizer series has done great things with delays, and the Orville continues that tradition. Stereo 3-D CircleDly is one of my favorite patches. It's a psychedelic panning and phase-shifting stereo delay that creates the illusion of going in and out of the speakers. The effect is not unlike the classic Dynotronics Cyclosonic Panner that was popular in the '80s. It's not a 4-channel patch like some of the Orville's Surround patches, but Stereo 3-D CircleDly goes a long way toward creating a dimension that seems to exceed the stereo field. Surround sound is one area that clearly figured heavily in the Orville's design; the unit offers a substantial number of 4-channel patches.

The Orville sounds great and has a warm, smooth, and clear sound that takes advantage of its impressive processing capabilities. Because the Orville offers so many types of effects, it's hard to pick a favorite, but I found myself using the reverbs and delays more than the other factory programs. The Orville handles effects such as distortion and sonic degradation (using bit-depth reduction, for example) very well, so I used those programs a lot too.

Combining the A and B processors to create one massive effects program can yield spectacular results if the right programs are chosen. I tried many combinations, and among the most successful were distortion into delay or reverb, delay and reverb in parallel, sampler into delay or reverb (for triggered processed samples), filtering into delay or reverb, and distortion into filtering.

WITH THE PROGRAM

Programming the Orville is conceptually simple, but the process is actually quite involved. The general paradigm is similar to that of a modular synthesizer:

FIG. 3: The Orville's optional remote control unit duplicates the Orville's front-panel controls and adds eight rotary controls.

modules (in this case, DSP modules) are connected to create a final program (patch). Much of the complexity arises from the Orville's large number of modules. The module categories include Bridge, Control Math, Control Process, Delay, Detector, Dynamic, External, Filter, Interface, Math, Miscellaneous, Mixer, Node, Oscillator, Pitchshift, and Reverb. That substantial list yields 167 modules, and although it isn't necessary to learn them all in-depth, becoming familiar with the programming process (and the basic modules' functions) takes time.

Fortunately, Eventide offers a freeware editor for Windows that has been available since the days of the DSP4000. The program, called *VSigFile,* has been updated to support the Orville and can learn new module definitions as the unit is updated and new modules are developed. Hardly elegant or pretty, *VSigFile* is an absolute necessity for serious programming. It lets you view and edit module parameters, and you can use it to transfer patches to or from the Orville if you want to modify an existing patch. *VSigFile* also lets you build programs from scratch and upload them from your PC to the Orville.

Dealing with the interface setup is arguably one of the thorniest aspects of creating patches with the Orville. That's because an Orville patch's interface elements—onscreen knobs, soft-key assignments, virtual meters, and so on—are modules that must be connected and configured down to the most minute detail (such as the

Orville Harmonizer Specifications

Sampling Rates	44.1, 48, 88.2, 96 kHz
Resolution	16- and 24-bit
Digital I/O	(4) AES/EBU or (2) AES and (2) S/PDIF
Analog Inputs	Any combination of (4) balanced XLR or (4) unbalanced mono ¼"
Analog Outputs	(4) balanced XLR
Signal-to-Noise Ratio	>110 dBA
Frequency Response (+0/-0.1 dB) @ sample rate	20 Hz–20 kHz@44.1 kHz; 20 Hz–22 kHz@48 kHz; 20 Hz–44 kHz@96 kHz
Sampling Time	174 sec. mono
Digital Delay Time	80 sec. mono (40 secs. per processor), or 214 sec. for DSP A using the 174 secs. of sampling time for delay
Remote Control Inputs	As many as (2) footpedals or (6) footswitches or a combination
Dimensions	2U × 12.5" (D)
Weight	12 lbs.

increments in which a knob moves). Fortunately, you can look at existing Orville patches and reuse individual modules or even whole groups of modules in your patch-creation process; you don't have to start from zero every time. In fact, as you develop a patch library, you'll accumulate commonly used sets of controls and processing blocks that can easily be reused.

Connection to the *VSigFile* program is done through MIDI or the Orville's built-in RS-232 serial port, which I used during the review process. The connection couldn't be simpler—one serial cable and some menu selections, and I had bidirectional communication between the box and the PC. Everything worked as expected.

Another convenient aspect of the Orville's PC interface is its ability to receive operating-system updates through Flash ROM. An updating application can be downloaded from Eventide's Web site for free. I used that application to get the latest Orville operating system; it was a painless process that took less than ten minutes and required no chip swapping.

The Orville also provides a built-in RJ45 port, which connects to an optional hardware remote network called EVE/NET. You can connect as many as four remotes ($1,595 each) to as many as four Orvilles (in parallel), which allows for installation in a variety of studio situations. Connections are made with Ethernet cables (CAT-5). The remote duplicates the Orville's front-panel controls and adds eight soft rotary encoders (see **Fig. 3**). Those always address the display's first eight parameters, which are duplicated on the remote. Each rotary encoder has a built-in switch activated by pushing down the knob.

ENDLESS POSSIBILITIES

The Orville's documentation is divided into two main parts: the Operating Manual and the Programmer's Manual. The Operating Manual covers the basic functions, including audio connections, control layout, basic patch selection, and parameter editing. The Programmer's Manual explores the Orville's modules, offers patch-editing tutorials, and shows you how to create programs from scratch. The documentation is generally straightforward and clear, and the two manuals overlap very little.

The Orville was a long time in development, and that is apparent in its sonic integrity and great flexibility. If you're familiar with Eventide's products, you'll feel comfortable with the Orville's design. If you're new to the Eventide universe, it won't take long to become acquainted with the unit's basics, though the intricacies of patch editing take time to master. Nonetheless, the processor's sonic possibilities are truly world-class and should prove endlessly useful in just about any music or post-production situation. The Orville is a professional device, and it's priced accordingly; it proves the adage "You get what you pay for."

PRODUCT SUMMARY

Eventide
Orville Harmonizer
multi-effects processor
$5,695

FEATURES	4.5
EASE OF USE	3.5
AUDIO QUALITY	4.5
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Great sound. Four-channel operation. Remote capability. Extremely flexible dual-processor architecture. Software-upgradeable through Flash ROM. Patch creation with PC freeware application.

CONS: Expensive. Routing page is counterintuitive.

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Peter Freeman *is a freelance bassist, synthesist, and composer living in New York City. He has worked with Seal, Jon Hassell, John Cale, Nile Rodgers, and Shawn Colvin, among others.*

ROYER LABS

SF-1

A warmer, flatter ribbon with superb lows and an extended high end.

By Myles Boisen

Royer Labs is a small company dedicated to reviving ribbon microphones for the digital-recording era. I reviewed the company's flagship mic, the R-121, for the May 1999 issue of **EM**, and I was so impressed that I purchased one for my studio. Since then, that mic has acquired most-favored status at the studio for a number of applications, especially miking guitar amps and brass instruments. My colleagues were impressed, too, and honored the R-121 with an Editors' Choice award in 2000.

The inspiration for the SF-1 came from the Speiden SF-12, a stereo coincident ribbon microphone originally hand built by Bob Speiden but now manufactured by Royer (see **Fig. 1**). The SF-1 is the mono version of the SF-12—if you sawed an SF-12 in half and doubled the electronics, you would essentially have two SF-1s.

The SF-12 and SF-1 have thinner ribbons than the R-121 (1.8 µm compared with the 121's 2.5 µm) and different magnet structures. According to the manufacturer, those design aspects contribute to superior transient response (for which ribbon mics are already prized) and improved high-frequency response, albeit with increased fragility of the ribbon element itself. Of course, all ribbon microphones need to be shielded from powerful blasts of air and handled with care. The SF-1's output level is comparable to that of other ribbon mics, requiring 15 to 20 dB more gain than an average condenser mic.

Like its predecessors, the SF-1 comes in a beautifully crafted wood box with a nylon mic clip and a lifetime warranty to the original owner (for repair or replacement at Royer's option). The mic body is fashioned from ingot iron and has a matte black finish. Optional accessories include the Audio-Technica AT-84 shock-mount ($72), which Royer supplied for this review; a PS-100 metal-mesh pop filter ($47.50); and a Sonosax SX-M2 stereo mic preamp ($1,250), which Royer also supplied. Royer sells the preamp as an accessory because of its high-gain and low-noise characteristics, which are desirable for ribbon-mic use. The SX-M2 is a well-built, portable, and compact unit not much larger than a Sony Walkman. It provides 76 dB of gain and can be powered by battery or DC current. The SX-M2's performance lived up to its impressive specs, exhibiting very low noise and a somewhat smoother sound than my main test preamps.

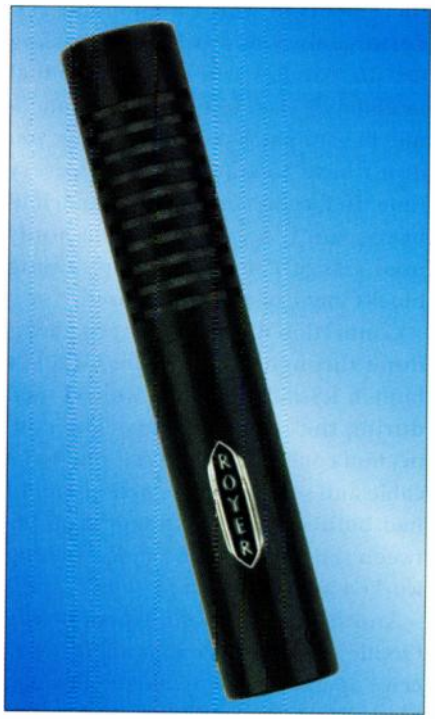


FIG. 1: The SF-1 is the second mono ribbon mic offered by Royer Labs. It has a warmer sound and flatter, more extended frequency response than the Royer R-121.

LINEUP

To get a sense of the SF-1's sonic character, I did some comparison testing. I set up a three-mic cluster: the SF-1, a Royer R-121, and alternately a Coles 4038 ribbon mic and an Oktava MC 012 small-diaphragm cardioid condenser mic. I used BLUE Kiwi microphone cables and Focusrite Green preamps and recorded to a Sony PCM-800 8-track digital recorder.

First, I performed a standard loudspeaker test. Although the R-121 did work some magic on the bass end of my boom box source, the SF-1 was more faithful to the source on a selection of mixes. The SF-1 highlighted ambience and percussive details pleasantly; after a number of listening passes, I rated the SF-1 tracks as the best of the group.

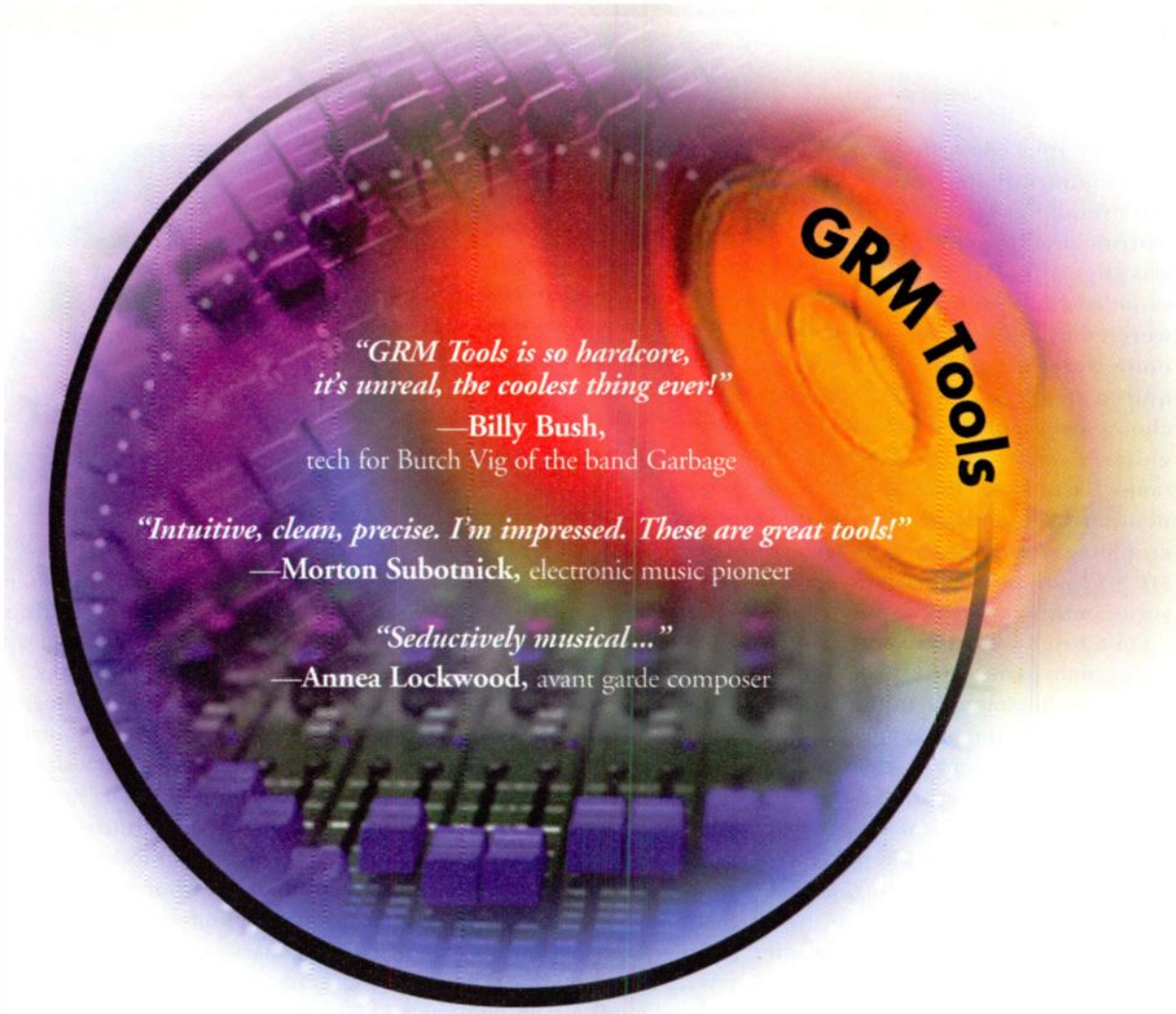
Frequency charts posted by the manufacturer show the SF-1's response deviating less than 3 dB from 40 Hz to 15 kHz; the response is flatter than the R-121's, which a comparison of the two charts makes clear (see **Fig. 2**). Some EQ sweeps confirmed the flatness of the SF-1's frequency response—it was easy to hear the effects of boosting in sweeps throughout the bass and treble ranges, and there were no evident weak or dead spots. However, in this test and on instrument tracks, I noticed a bump in the SF-1's response around 200 Hz.

Next, I compared the SF-1's off-axis response to that of the other mics by positioning the boom box several feet away at about a 90-degree angle to the mic cluster's side. Impressively, the SF-1 retained solid low-end characteristics that were lost on the R-121 and the small condenser. Both mics also emphasized unusual timbres in the source, indicating a less-than-flat response, and sounded thin or diffuse.

Last, I compared the two SF-1s to hear how similar they sounded. For the sake of thoroughness, I used three solid-state preamps: the Sonosax SX-M2, a Focusrite Green, and a Sytek MPX-4. I perceived only a minor difference in the high-end "air" between the pair of SF-1s. Other than that, the microphones sounded identical and perfectly matched.

STRINGS ATTACHED

For perspective, I left the mic cluster intact for the instrument tests. First up was an acoustic guitar strummed



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and fingerpicked at close range. The SF-1 sounded flatter than the R-121 but overall was less flattering to the instrument: although the high end was slightly clearer, the bass notes sounded indistinct. The Royer mics were comparable in terms of self-noise (very low) and preamp gain required (about +55 dB). If I had to choose between the two on acoustic guitar, I would opt for the R-121's tangy flavor over the SF-1's relative neutrality. However, of the three microphones in the cluster, the Oktava MC 012 condenser mic was my favorite in this application.

On a bright, close-miked acoustic slide-guitar part, the SF-1's high end was closer in character to the Oktava 012's. But there the ribbon mic's softer sound was an advantage, despite the duller-sounding upper range and the somewhat boomy and unfocused lows. With both mics strapped in the AT-84 suspension shock-mounts, the SF-1 proved to be more immune to foot stomping and stand-borne vibration than the R-121.

At a distance of two feet, the SF-1 provided a full low end and realistic highs for an assortment of guitar styles. A cut at 220 Hz took out a slight boominess in the sound and let my inexpensive Hyundai guitar sound much richer than it did through the other microphones. Not surprisingly, in the high end above 8 kHz, there was still no comparison with the crisp timbre of the Oktava condenser.

During a session with a Martin Backpacker mandolin, the SF-1 added a warm, supportive character. The performer, guitarist Michael Bizar, praised the microphone's qualities, and I agreed that the SF-1 warmed up this instrument very nicely.

SF-1 Specifications

Acoustic Operating Principle	electrodynamic pressure gradient
Generating Element	1.8 µm aluminum ribbon
Polar Pattern	bidirectional (figure-8)
Frequency Response	30 Hz–15 kHz (± 3 dB)
Sensitivity	–52 dBV (1v/Pa ±1 dB)
Output Impedance	300Ω @ 1 kHz (nominal)
Maximum SPL	>130 dB
Dimensions	5.6" (L) × 1" (W)
Weight	9.3 oz.

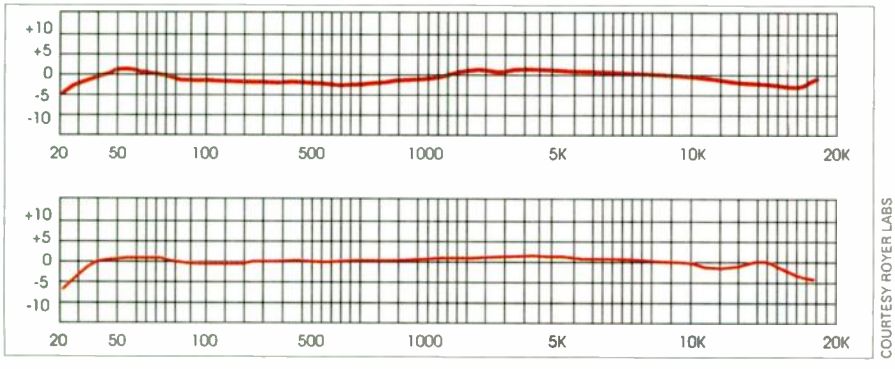
ELECTRIFYING

Because the R-121 has become a first-pick microphone for recording electric guitar at my studio, I was eager to see if the SF-1 could match or perhaps even beat it. It didn't. Positioned about two feet from a cranked Fender tube amp with two 12-inch speakers, the SF-1 conveyed a hollow midrange and too much low-end mud. A 4 dB boost at 2.5 kHz brought the SF-1 closer to the R-121's character, but the SF-1 still didn't have the presence and ready-to-rock tone that have made the R-121 so popular among guitarists and engineers.

Likewise, on a mellow, amplified jazz guitar, the SF-1 was murky sounding compared to the other ribbons. In this application, its extended high end brought out some amp noise but no special qualities. As much as I appreciate the SF-1's virtues as a flat and extremely warm mic, for electric guitar I usually want a mic with some attitude. The SF-1 has a neutral quality that makes the R-121 and Coles 4038 seem aggressive in comparison.

On the other hand, during a session with guitarist John Shiurba, the SF-1 worked wonders on a small, solid-state Vox guitar amp. It managed to add punch and authority yet smooth out rough edges on the challenging array of textures Shiurba created. I also tried the SF-1 directly on another of Shiurba's guitars, an unamplified hollow body, as he played in the same room with other musicians. This time the mic provided incisive high-end detail. Also, the mic's off-axis pickup, which was readily apparent when I brought up the acoustic instrument track in the ensemble setting, was remarkably uncolored. Often during multiple-mic sessions, bleed from condenser mics creates muddiness, unwanted room sound, and narrow-band coloration throughout the frequency range. But in this case, the leakage was not a problem, especially after I equalized some "thump" out of the guitar with a 4 dB cut at 200 Hz.

I experimented with the SF-1 on two keyboard-and-amp rigs (solid state) as the players ran the gamut of samples, synth patches, and industrial noise. During tracking and mixing, I was impressed by the SF-1's immediacy and its sympathetic treatment of pure synth tones, high-resonance peaks, and low-end material in the 40 to 80 Hz range. I didn't compare any other mics during this session, and I didn't feel compelled to, either. Under demanding conditions the SF-1 reproduced diverse, full-frequency sources perfectly, with no



COURTESY ROYER LABS

FIG. 2: The SF-1's frequency plot (bottom) is clearly flatter than the R-121's (top).

harshness or dulling. In the mix, the tracks needed minimal EQ—just a small amount of 4 to 8 kHz sweetening or the occasional low- or upper-midrange cut around 1 kHz.

Brass instruments are a traditional favorite for ribbon miking. I tried the SF-1 on trumpet and received great results. Despite some raspiness in the high end, the SF-1 gave the trumpet a bigger sound and an enhanced sense of low-end air movement compared with the R-121 and Coles 4038 ribbon mics. On open horn, the Coles 4038 had less fizzle, but the SF-1 displayed a touch more warmth. On Harmon-muted trumpet, the SF-1's overall response was a real boon, supplying clear, high-harmonic richness and an authoritative low end.

SHAKE, RATTLE, AND ROLL

The SF-1 improves markedly on the R-121's already formidable capabilities as a percussion mic. On tambourine, the R-121 sounded dull and crunchy in comparison, and the small-diaphragm condenser sounded a bit too bright and piercing. The SF-1 rendered the tambourine in a detailed and surprisingly listenable way; indeed, it sounded superior to any ribbon mic I've tried in that application.

On shaker, though, the Oktava MC 012 was my favorite, and the SF-1 offered obvious improvements on the R-121 in terms of high-end pickup and transparency. Yet on jingling keys, miked from two feet away, the SF-1 proved much more accurate than the condenser microphone. Interestingly, when I moved the keys to about one foot from the mic cluster, the SF-1 sounded almost identical to the Oktava 012. For that source sound, I found the condenser's slight high-end advantage to be a sonic disadvantage, because it provided too much stimulation to my middle-aged cilia. The SF-1 is truly the first ribbon microphone I have ever considered for delicate high-end percussion duties, and I looked forward to trying it on drums.

Fortunately, fellow engineer Karen Stackpole was reviewing an Ayotte drum set for *Onstage* (**EM**'s sister publication), so I was able to evaluate the SF-1 pair in XY coincident and split-overhead configurations. At approximately seven feet above the floor, the XY and spaced arrangements of the SF-1 pair underrepresented the cymbals and sounded too tubby. Compared with the Oktava 012 XY pair, the SF-1 grabbed a great snare sound and painted a much more robust and immediate picture of the kit's drum portion. But even with EQ, the cymbal sound was just too dull.

Remembering my experience with the key test, I moved the split-overhead SF-1s closer—nearly on top of the left and right cymbal clusters—so that no cymbal was more than three feet from the mics. Suddenly, the sound of the drums snapped into focus: not only was there much more detail from the cymbals but also a remarkable combination

PRODUCT SUMMARY

Royer Labs
SF-1
ribbon microphone
$1,075

AUDIO QUALITY	4.5
VALUE	5.0

RATING PRODUCTS FROM 1 TO 5

PROS: Flat frequency response. Better upper high-end response than most ribbon mics. Superb bass response and punch. Great for some small percussion. Viable for drum set. Excellent on keyboard amps and brass instruments. Remarkably uncolored off-axis response. Low self-noise. Beautifully crafted wood case. Lifetime warranty to original owner.

CONS: As with most ribbon mics, attention to placement and EQ required for best results. Increased fragility due to thin ribbon. Prominent boost around 200 Hz noted on many sources. Low-end response can sound murky or unfocused on some acoustic and electric guitars.

Manufacturer
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e-mail sales@royerlabs.com
Web www.royerlabs.com

of punch and clarity also emerged from the floor tom and snare. With a little high-end boost, that setup could provide a big and distinctive sound for jazz or funk recording.

I also experimented with the SF-1 as a mono drum overhead, comparing it with a single Oktava 012. Placed above the center of the kit and about five feet from the floor, the SF-1 picked up viable cymbal sounds and captured a huge snare tone that sounded much truer than what the condenser mic captured.

When raised two feet higher, the single SF-1 produced a sound surprisingly comparable to the Oktava 012. With a high-frequency shelving boost of +3 dB and a broad low-end cut of 3 dB at 200 Hz, the SF-1 nearly matched the small-diaphragm mic's crispness and provided a butt-kicking low end to boot. It took a little getting used to, but toward the end of the listening session, I really began to like the full sound emerging from the SF-1, and I decided to experiment with it further in upcoming sessions.

During the listening evaluation, Stackpole noted that the SF-1 sounded "better and brighter when the mics got closer, and they did seem to give the drums some beef." Although in the end Stackpole preferred the drum sound captured by the small-diaphragm condenser pair, she remarked that "the Royers are definitely usable—not a bad sound, just darker."

NEW VOICE

As a frequent and enthusiastic user of the Royer R-121, I approached this review with great curiosity. Could the SF-1 actually improve upon the R-121's upper high-end response? The answer is an unequivocal and resounding yes. Is the SF-1 a great ribbon mic? Once again, the answer is yes. As I learned from my tests, the SF-1 does some things better than other ribbon mics (including the R-121). But not surprisingly, there are also things it doesn't do as well.

I was impressed by the SF-1's potential as a percussion and drum mic. With proper placement and a few EQ nudges, its clarity rivaled a small-diaphragm condenser mic that I regularly employ. The SF-1 also sounded wonderful on trumpet, and I would expect it to perform as well or better on other members of the brass family. In addition, I was literally moved by the superb bass response and punch of the SF-1—it reproduced powerful low-end air movement in a way that only a few high-end condensers can.

With this new entry into the underpopulated world of studio-grade ribbon mics, Royer has created yet another distinctive, versatile, and great-sounding microphone. Not only is it an excellent complement to the R-121, but the Royer Labs' SF-1 also has a voice and capabilities all its own.

U&I SOFTWARE

METASYNTH STUDIO 2.7 (MAC)

Unique visual synthesis brings out the sound designer in anyone.

By Jeff Burger

Since it burst onto the scene in 1998, U&I Software's *MetaSynth* has offered Mac users a unique collection of sound-design and composition tools. **EM**'s last review of the program (in the August 1998 issue) covered version 2.0, and since that time, a slew of new features has been added. U&I now offers the MetaSynth Studio 2.7 bundle, which consists of sibling products *MetaTrack* 1.4 and *Xx* 1.3 along with updated *MetaSynth* software. You can also purchase the MetaSynth Studio 2.7 bundle's elements separately. *MetaSynth* is $299. *MetaTrack* ($99) and *Xx* ($129) are available only through the Internet.

With its algorithmic composition, MIDI-to-PICT, and PICT-to-MIDI features, *Xx* hasn't changed much since its review appeared in the March 1999 issue. The same can't be said about *MetaSynth*, which has evolved considerably in the past few years. Also, *MetaTrack*, a multitrack environment for arranging and scheduling *MetaSynth* sounds, now augments *MetaSynth*. (For more information about *MetaSynth*, see "Master Class: Mastering *MetaSynth*" in the February 2001 issue.)

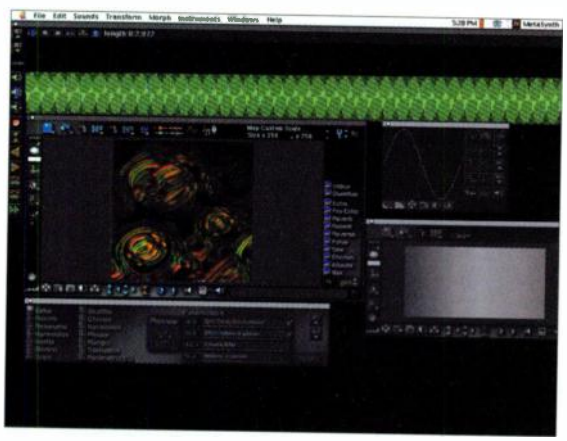


FIG. 1: *MetaSynth*'s main work areas are divided by task. The Image Synth is pictured here at mid left, and the Sample Editor appears above it. The remaining work areas are the Effects Palette (bottom left), the Wave Table Synth (middle right), and the Effects Palette (bottom right).

META BASICS

MetaSynth lets you design, manipulate, and render an unparalleled array of sounds using tools borrowed from the world of computer graphics. At the program's core is the Image Synth, a two-dimensional canvas that contains the information used to play any sound source you select. The Image Synth's horizontal axis is time, and its vertical axis is pitch (see **Fig. 1**). You can import photos and graphics, paint directly on the canvas with a variety of brushes designed for musical applications, and apply image-processing effects à la Adobe *Photoshop*.

Each pixel in *MetaSynth* represents one of 1,024 oscillators, with brightness controlling volume and color determining pan position. The Filter Palette provides the same tools as the Image Synth but acts as a 128-band, time-variant, dynamic filter for sounds loaded into *MetaSynth*'s Sample Editor. You can receive wildly varying results from a single picture by changing the harmonic scale associated with the vertical axis, the timeline's duration, or the sound source selected for the oscillators. The results can be anything from new instruments to techno sequences to otherworldly soundscapes.

The Sample Editor is also powerful. Like the waveform editors of many audio applications, the Sample Editor provides standard cutting, copying, and pasting of mono or stereo audio files and offers many additional processing features such as morphing and convolution. The final major work area is the Effects Palette, in which you can apply a number of effects to a sound and manipulate their parameter settings in real time.

Minimum System Requirements

MetaSynth Studio

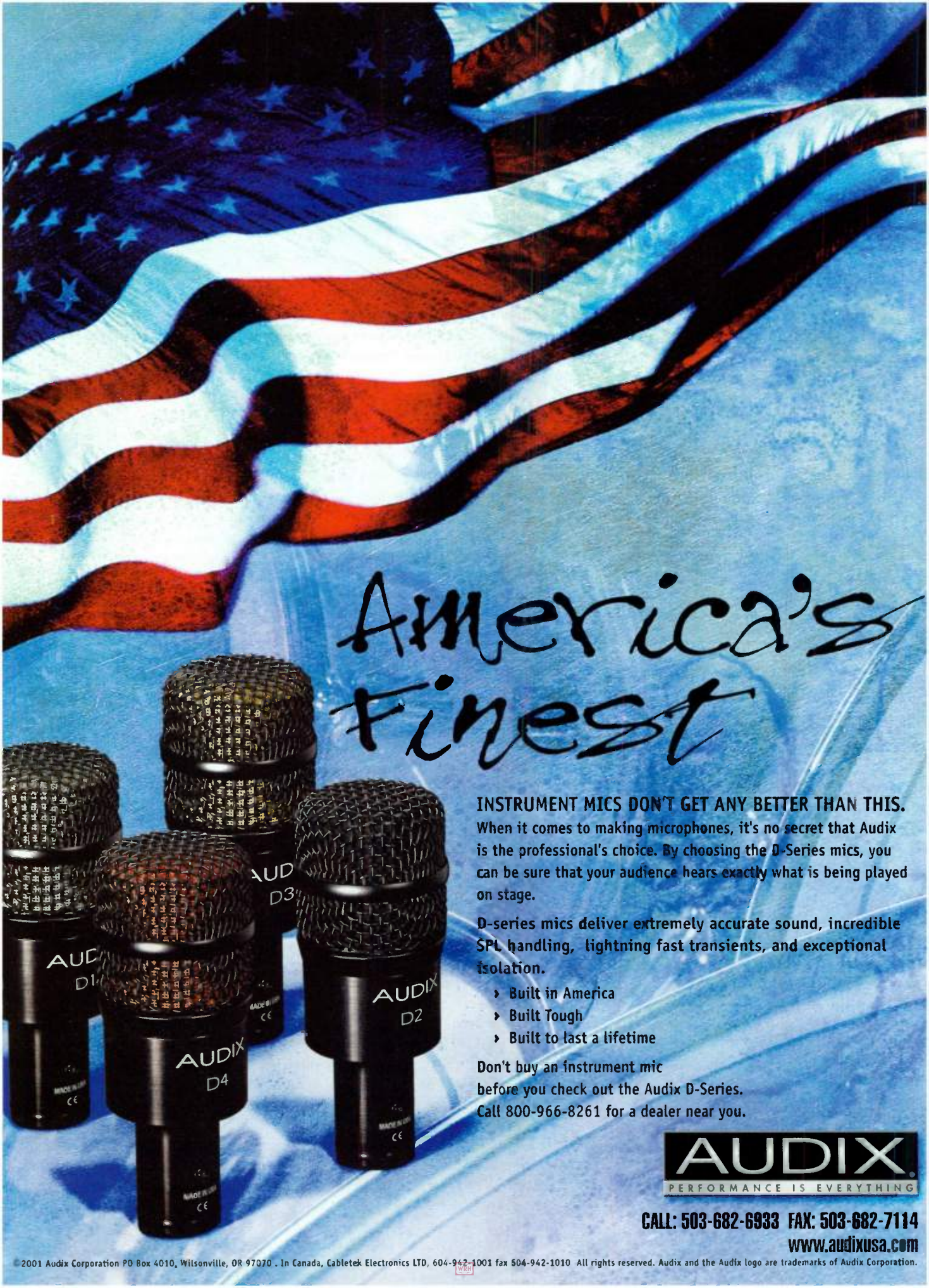
Power Macintosh 601/120; 16 MB RAM; Mac OS 7.3; *Sound Manager* 3.1; *QuickTime* 2.0

NEW SONIC TOOLS

One of *MetaSynth*'s more distinctive new features is the ability to use displacement maps. A displacement map uses its pixels to displace or warp the master image's pixels. The luminosity of each pixel in the map determines how much the corresponding pixels will move in the master image—50 percent gray produces no change, and black and white represent extreme displacement in either direction. As with most other aspects of *MetaSynth*, you can use preset images or import any PICT file or clipboard image as your map.

You can apply displacement maps in a variety of ways, all of which are guaranteed to result in outrageous images and equally outrageous sounds. For example, you can displace only one axis or use the Smoothing option to produce continuous curves in places where line breaks would normally appear. Smoothing is appropriate for modulation effects in the Image Synth or sweeps in the Filter palette. When you want to alter a sound's harmonic structure, transpose melodies, or create rhythmic effects, it's better to turn Smoothing off.

Other new tools let you work at the spectrum level. The Instant Spectrum command creates a high-quality fast Fourier transform (FFT) spectrum analysis, which analyzes as many as 1,024 partials from the first 2,048 sound samples loaded in the Sample Editor. You can use the analysis as a source for the Synthesize Spectrum feature. The results are not dynamic, so you have to apply a filter or other tools to make the resulting sound move through time. Instant Spectrum's most common



application is to assign reference pitches to the results of Synthesize Spectrum, so you can create either single- or multisampled input sources for the Image Synth or for use in a traditional sampler.

A spectrum analysis can also yield interesting sounds when it is used as a custom scale for the Image Synth. Alternatively, you can employ the spectrum as a filter. Processing a rich sound like a horn's with a spectrum from another rich instrument like a cello, for example, can render sounds suggesting a stringy horn.

The new Wave Shaping feature uses a standard *MetaSynth* envelope to selectively remap the amplitudes of individual samples of the sound loaded in the Sample Editor (see **Fig. 2**). The graph represents an input-to-output map (or *transfer function*) across the range of sample amplitudes, so a 45-degree line, for example, represents a linear response or no change. Applying this tool to various waveforms is a great way to explore new sounds. Use it to impart a subtle distortion or a denaturing quality to a clean sound. If you plan to add dynamic filtering, the extreme settings yield a bevy of rude harmonics that may aptly suit your needs. As you would expect, the wave shaper includes a number of tools that let you create custom curves. The new Remap Colors feature manipulates an image's color in ways that are similar to Wave Shaping's effect on amplitude.

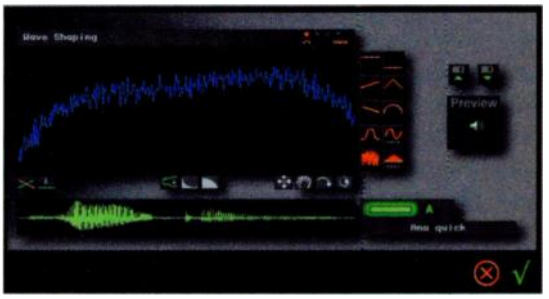


FIG. 2: The Wave Shaping tool is one of *MetaSynth*'s most powerful new features. The envelope in the middle of the screen represents the transfer function, which is used to map incoming amplitude values to outgoing values.

Two new input sources for the Image Synth are available: Granular and Sample Granular. Unlike the Grain option in the Effects palette, which provides a number of parameters to adjust the granular effect, those new features apply a preset granulation algorithm to the Wave Table or current Sample Editor contents. They work by taking small snippets of the source sound and looping them with slight overlaps. The frequency of the overlaps adds harmonics, with results similar to those of a tunable resonant filter. The amount and nature of the new harmonics are automatically derived from the Image Synth picture. Without any adjustable parameters, some experimentation is required to arrive at an image that yields usable results.

MORE META GOODIES

MetaSynth 2.7 can import and export custom scales of 1,024 values. That is quite significant when you consider that *MetaSynth* employs scales for harmonic spectra and melodies. The *MetaSynth Studio* CD includes a vast selection of custom scales (approximately 4,000, according to the manufacturer), and you can also create scales from the output of the Instant Spectrum feature. If those don't manage to keep you busy, you can create scales manually, either by entering text or by using Kenneth Newby's *ScaleComposer* freeware (which is available at the U&I Web site) to create them algorithmically.

For version 2.7, U&I shook out all known bugs and improved *MetaSynth*'s overall efficiency. The silent blue grid now has a features submenu that makes creating rhythmic and note grids easier. The Grain effect computes more accurately and is two to three times faster, making it practical for lengthening sounds and transforming mono to stereo. In the past, *MetaSynth* Instruments remained in memory until deleted; the program now swaps them into RAM only when in use. The absolute sample-size limitation of the Sample Editor was removed as well. You can also save several steps when editing *MetaSynth* presets from *MetaTrack* by hot linking through AppleEvents.

METATRACK 1.4

MetaTrack is a rudimentary, 16-track sequencer designed for mixing *MetaSynth* sounds; its operation is straightforward. You begin a session by opening a library of presets that you created with *MetaSynth* into a source palette in *MetaTrack*. Then you create sonic montages by simply dragging various sounds from the palette onto a grid that shows time and tracks. Basic tools facilitate quantization, fade in and out, insertion and deletion of measures, and a link back to *MetaSynth* for editing.

MetaTrack has buttons for muting and applying effects to tracks, volume and pan settings for tracks, and global volume and pan settings. Although the design is not as intuitive or informative as a traditional multichannel mixing display, it does make it possible for all of the controls to appear on a single screen.

MetaTrack provides nearly a dozen effects, including EQ, compressor, rotary speaker, delay, and early reflections. The quality of the effects is decent, especially considering that a moderately powered computer gives you an

PRODUCT SUMMARY

U&I SOFTWARE

MetaSynth Studio 2.7 (Mac)
sound-design software bundle
$429

FEATURES	4.0
AUDIO QUALITY	4.0
EASE OF USE	3.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Unique synthesis options. Great sound quality. Well-integrated suite of tools.

CONS: Exploiting potential requires time commitment. Nonstandard interface sometimes not intuitive.

Manufacturer

U&I Software
tel. (800) 811-1991
e-mail order@uisoftware.com
Web www.uisoftware.com

effect on each of the 16 tracks simultaneously. The program also lets you load and save effects settings.

Once a *MetaTrack* mix is perfected, you can render it as a Sound Designer II file (interleaved or dual mono). You can also export tracks if, for example, you want to process them further in another application. Before rendering, you can apply a selection of reverbs to the mix. Despite the fact that *MetaTrack* is a bare-bones program, it offers an easy, powerful, and inexpensive method to create complex soundscapes using your *MetaSynth* discoveries.

META THOUGHTS

U&I Software has produced a set of tools that is completely unique in computer music. Mastering those new tools takes a while, but the results are worth the time invested. Discovering new sounds with *MetaSynth* is so much fun that it's addictive. *MetaSynth* is a must-have tool for anyone doing sound design or wishing to explore the realm of abstract sound. It is also great fuel for the imagination in the techno and dance music genres.

▼

You can paint with a variety of brushes designed for musical applications.

The *MetaSynth Studio* bundle bridges *MetaSynth*'s unique visual metaphor with more traditional applications. Although you could simply dump multiple *MetaSynth* sounds into a multitrack audio editor, *MetaTrack* is ready-made for that task. *Xx* is great for generating new melodic phrases and even complete scores, and it offers a way to exchange data with *MetaSynth*. The threesome yields an incredible recursive loop of sonic-processing and composition tools capable of rendering final mixes for many musical applications. The bundle's 300 MB of additional samples take *MetaSynth* well beyond the ordinary. ❁

Quick Picks

ELAB

Abstract HipHop

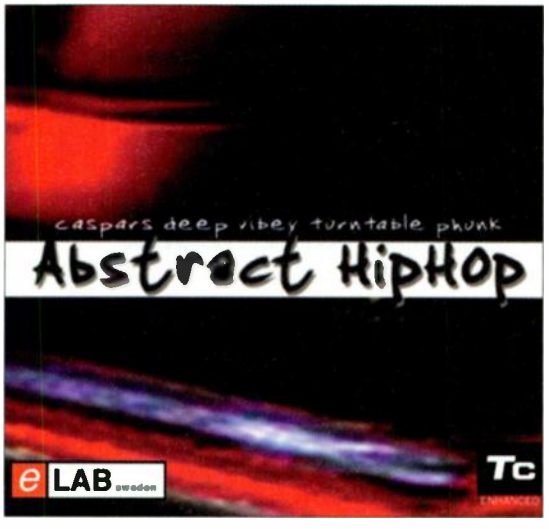
By Jeff Obee

Face it: there are hundreds of hip-hop construction kits on the market. What distinguishes eLab's collection as abstract? Is it so different from the teeming multitudes of breakbeats and loops?

Abstract HipHop ($99.95) comes with two discs: a standard audio CD and a CD-ROM of WAV files prepared for Sonic Foundry's *Acid*. The CD-ROM contains discrete left and right mono files and complete stereo tracks. Although the audio CD doesn't provide you with access to mono files, the CD-ROM's individual mono files played in tandem provide a broad, full-stereo image consisting of two distinct but musically compatible loops playing in sync.

Let's Chill

The collection's subtitle is *Caspar's Deep Vibey Turntable Phunk*. Indeed, there's a distinct and appealing "chill" vibe to the package's loops. The disc's grooves sit at a relaxed 90 or 100 bpm, so everything is readily interchangeable.



ELab's *Abstract HipHop* sample CD features a mix-and-match collection of diverse, rhythmically compatible grooves.

The CD-ROM menu presents six categories, neatly organized into folders by bpm. The Inspiration Kits are full construction sets, each with roughly half a dozen compatible drum loops and a variety of instrument grooves provided by electric pianos, vocals, sound effects, bass, guitar, and horn.

The LoopTools category features drum grooves different from those in the Inspiration Kits. In both categories, the producers nailed the beats. You get soulful, analog-infused two- or four-bar patterns at every turn. Many patterns are laden with percussion. They range from sweet grooves in the style of Marvin Gaye's "What's Goin' On" to unusual takes on Brazilian- and bossa nova–influenced rhythms to down-and-dirty street funk.

Smooth Grooves

The Music Loops and Bass Loops provide a few smooth and solid bass lines but, overall, lean heavily toward electric piano and guitar licks and lines. Piano and guitar provide enough harmonic variations to give you room to move. The Sounds, Chords, and Riffs category complements the Music Loops beautifully with soul sax, flute, and horn sounds and riffs; more guitar and electric piano pieces; synthlike string pads and stabs; and a few instruments in combination.

The Vox and FX selections provide R&B and gospel stylings; spoken clips; squelchy, lo-fi–radio vocal effects; and breathy female-vocal trinkets. Synthetic crickets, reversed cymbals, and synth noises fill out the FX section. Xclusive Drums contains 173 quality vintage–drum-kit, beatbox, and percussion one-shots that include kicks, congas, snares, hi-hats, toms, rims, claps, and shakers.

Many sounds are drenched in delay and reverb, creating an ambient atmosphere. At times the delay is a bit overdone. That isn't necessarily a criticism, but such effects are sometimes better used in moderation and left to the end-user's discretion.

I occasionally heard the remnants of the drum loops with some other sounds, probably because much of the material was extracted from reel-to-reel tape. It wasn't a distraction, and besides, the samples benefit from the use of magnetic tape. There is a pleasing, warm analog sound overall.

The Abstract Truth

The CD-ROM covers the bases, so why include an audio CD? The audio CD serves as a backup disc and lets you audition files during a session while the CD-ROM is in use. Computer-phobic artists can use the audio CD with their vintage gear, and performing DJs might also find it useful.

You won't be thrilled with the audio CD's documentation, however. It lacks indexing and file times for each sound in a track. The cardboard packaging is hip, but one of the plastic disc holders inside came unglued after a few uses, making it unacceptable for hauling to performances.

The title's *abstract* is a tad misleading. Before I listened to the CDs, I thought the material might push the envelope with the use of esoteric plug-ins and sound-shaping applications. *Deep vibey turntable phunk* is a more accurate description.

Despite its title, though, the collection offers a lot of juicy material that distinguishes it from the pack. I can recommend *Abstract HipHop* on the strength of its drum loops alone; most grooves are superb and sit in a delicious pocket. Add a bevy of sound effects, music loops, and individual drum and percussion hits—all presented in formats to fit almost anyone's working style—and *Abstract HipHop* is an excellent purchase.

Overall EM Rating (1 through 5): 4

eLab/Big Fish Audio (distributor); tel. (800) 717-FISH or (818) 768-6115; e-mail info@e-lab.se; Web www.e-lab.se

RIOT ACT

Slow Death by Vise

By Jeff Obee

Slow Death by Vise ($79.95, audio CD; $99.95, AIFF/WAV CD-ROM) offers 68 tracks of deliberately twisted, rude, distorted, and unusual sound effects. The samples range in length from less than one second to more than two minutes. The collection owes its

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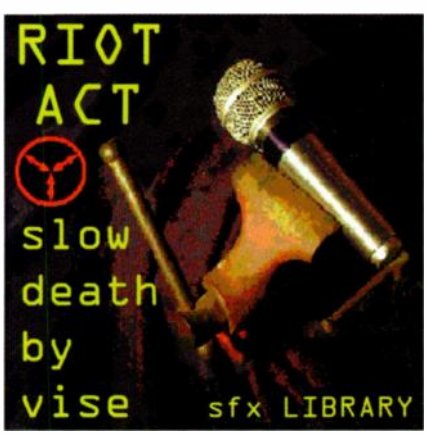
rough character to the lo-fi microphones and speakers used in the recording and to unconventional recording techniques.

The producers gathered plastic karaoke mics, old paging microphones, and piezo transducers and connected them to deliberately distorted and damaged speakers, car speakers, and Sony Walkman headphones. The producers then attacked those devices with a variety of implements and recorded the resulting signals. In addition, a Shure SM81 condenser microphone, which was pointed at the source from a foot away, served as an ambient mic. The combined signals were then recorded to separate tracks using Digidesign's *Sound Designer II*. The miked track, the ambient track, or an occasional blend of the two tracks made it to the final mix. Editing and mastering was accomplished with *Sound Designer* and Pro Tools.

Drop in the Bucket

When I first listened to the CD, I was momentarily taken aback. I thought that something was horribly awry in my studio. It turns out that the CD's initial sounds are 60 kHz noise bursts followed by a mic being tapped and a patch cord's unplugged end being touched. Beware: the documentation warns of drastic variations in volume levels from sound to sound.

The effects were created with a diverse selection of items and methods. A few offerings include a plastic mic recording while on fire, pop cans being crunched, a mic resting against a grinding wheel, and plastic and glass being hammered and crushed. The producers also used noises from items they dropped and rolled, including electric motors, a wooden rat trap, a condenser mic dragged on concrete, a microphone tapped or blown, a bottle rocket fired through a plastic tube, and a jar dropped in a bucket.



Riot Act's *Slow Death by Vise* provides unusual, raw sound effects captured from the signals of brutalized, cheap microphones and abused speakers.

The Drill Bit

Each track offers several takes of a specific sound event recorded with different microphone configurations. For instance, track 18, titled "Glass Jar Dropped in a Bucket," provides six takes: three takes of a condenser mic inside the bucket, which recorded multiple bounces of the jar, and three takes from the perspective of an ambient microphone above the bucket.

Track 5, "Attacked by Drills," offers six metallic grating sounds that run the gamut from "heavy rumble with medium-pitched screechy attack and fuzz decay" to "sustained harsh screechy electric motor." You're into extreme sounds, you say? Try "Drill Press Penetration," in which a rotating drill bit is pushed into a mic capsule until it hits the circuitry.

Slow Death by Vise features a large number of metallic sounds. Track 45, "Shaking Metal Box of Metal Parts," has a dozen sounds. The producers miked the box inside and outside. The shaking rhythm varies, as does the timbre. Recordings of automobile tow chains constitute 37 sounds on three tracks. The chains are dropped as a wad onto a mic and as a pile into a box, and are then gathered from the box.

The CD's sleeve provides an overview of the collection's contents. Comprehensive documentation is included on the disc in PDF and *FileMaker* formats. Sounds are listed by track with the sound title, description, track number, start point, and length. If you have *FileMaker*, you have that information and a card-view library, keywords, and user comments.

Word to the Vise

Riot Act aimed the CD at sound-effects designers working in video games and Foley for science fiction and horror TV shows and films. Unless you need raw, intense soundstage aural events, *Slow Death by Vise* probably isn't for you. However, if you work in a genre that requires unusual sounds, it's a resource you may want to have. Spend the additional $20 for the CD-ROM with WAV and AIFF files. You can download a free demo from Riot Act's Web site. Give it a listen.

Overall EM Rating (1 through 5): 3.5
Riot Act; tel. (708) 222-9842; e-mail feedback@riot-act.com; Web www.riot-act.com

Endlessflow Productions' *Ugly Remnants* sample CD for Sonic Foundry's *Acid* is a wonderful and weird collection of loops, pads, motifs, and more.

ENDLESSFLOW PRODUCTIONS

Ugly Remnants (Win)

By Julian Colbeck

Sonic Foundry's *Acid* loop-editing software has done well with its burgeoning library of grooves and effects. Nonetheless, there's always room for more effects, especially when the content is as fresh and unexpected as Endlessflow Productions' *Ugly Remnants* ($79.95).

Ugly Remnants is a truly massive, uniformly well-recorded collection of material in *Acid*-friendly WAV format. The material spans two CDs and is neatly categorized into folders (Bass, Beats, Beds, FX, Melodies, and so on) and subfolders. For example, the Beats folder contains drum loops, fills, percussion, rhythm effects, and vocalizations. Within *Acid*, you can instantly load or preview items, at which point you also see tempo, duration, and root key (where applicable).

Monstrous Sounds

If you can tell a lot about people by the samples they keep, then Mark Harbst, president of Endlessflow Productions, must be a science fiction and monster-movie buff. He's probably a fan of Anne Rice's novels, too, and perhaps he was a goth in years gone by. You get lots of scary guitar, twisted riffs, and ghostly wails and effects. The CDs boast some splendid distorted metal bass grooves, vocoded basslines, and such. *Ugly Remnants*' dark, eerie pads and drones positively ooze atmosphere.

The collection embraces drum loops, pads, motifs, sound effects, beds, and melodies played on every imaginable instrument, including koto, tuned percussion, and vocals. Stylistically, *Ugly Remnants* is an appropriate, if somewhat self-effacing, title; though samples are frequently laden with effects, they are not prettied up. The samples sound perfectly crisp and clear, but they are not designed for easy listening. Many are tailor-made for hip-hop and dance material, and you get a huge number that can be used in other types of music, from rock to jazz.

The Beats folder contents typify the product's exploratory feel. In addition to 151 beats in the Drum Loops category alone, disc 1 offers fills and a folder of percussion loops played with sampled tabla, real bongos, dumbeks, shakers, congas, and kitchen utensils. The Rhythm FX category contains industrial machinery, crunches, and munches. A vocalizations folder includes chants and monster noises. The rhythmic material is inspirational stuff—a far cry from the typical "me-too" dance drums on other sample CDs.

Behind the Curtain

The data is clearly organized on the CDs. However, the packaging consists of a grainy printout and generic CD labeling, with only the perpetrators' names supplied on the rear tray insert. That's a shame, because *Ugly Remnants* is an excellent body of work. Its sales could easily be hampered by the homespun physical presentation.

Harbst was the main force behind one of Sonic Foundry's most popular loop collections, *Universal Groove Elements,* and his knowledge of what works best in *Acid*-style compositions shines through this glittering collection. Although one or two song-form compositions speed you on your way, *Ugly Remnants* knits together well, and I don't classify myself as a power user of *Acid.* If you use *Acid* and hunger for more raw materials, *Ugly Remnants*' 1,900 loops—which span the weird and the wonderful—come highly recommended.

Overall EM Rating (1 through 5): 4.5

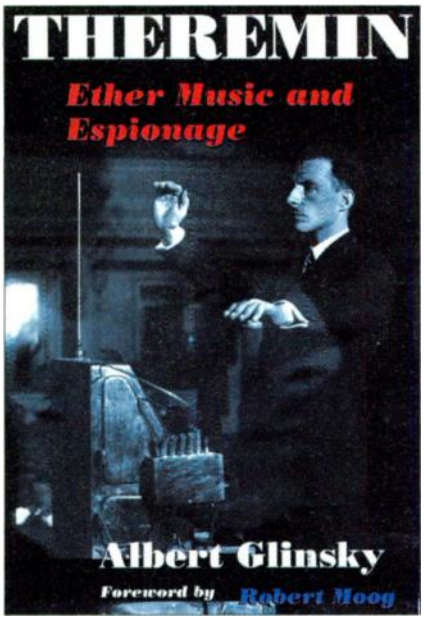
Endlessflow Productions; tel. (602) 404-9155; e-mail endlessflow@endlessflow.com; Web www.endlessflow.com

UNIVERSITY OF ILLINOIS PRESS

Theremin: Ether Music and Espionage

By Gino Robair

In electronic music's history, mystery and Cold War–related speculation have shrouded the life and work of one man: Leon Theremin. Albert Glinsky's biography *Theremin: Ether*



Albert Glinsky's *Theremin: Ether Music and Espionage* is the first detailed biography of Leon Theremin written in English.

Music and Espionage pieces together, for the first time in English, the facts about the inventor once known in his homeland as "the Russian Edison."

Glinsky creates spy novel–worthy suspense as he unravels the schemes Theremin and his associates used to cloak their work. The author explores a number of interesting themes, including Theremin's lifelong interest in reanimating the dead.

All Things Theremin

Lev Sergeyevich Termen was born in St. Petersburg, Russia, in 1896. As a young man, he excelled in science, engineering, and music. That interesting subject combination helped him—while still in his early 20s—to conceive the electronic instrument that would bear his name.

The theremin's success gave the inventor opportunities unheard of for Soviet citizens of the time. Theremin toured Europe promoting his instrument, which was as much a propaganda tool as he was. The concertizing led to a visit to New York City, where Theremin remained from 1927 to 1938. The revelations about Theremin's sudden departure from New York under mysterious circumstances and his whereabouts for several decades are what make the book most historically important.

As expected, Glinsky details the theremin's production and marketing. He also covers Theremin's many other inventions, including an early form of television, a polyphonic keyboard instrument, an aircraft altimeter, electronic security devices, and the bugging technologies used by the KGB.

In addition, the book documents Theremin's interaction with many important personalities of the 20th century. In the Soviet Union he gave a private demonstration of his "etherphone" to Soviet leader Vladimir Ilyich Lenin (who believed it would be "an ideal propaganda tool for electricity"). In the musical world, Theremin worked with composers Joseph Schillinger, Nicolas Slonimsky, Edgard Varèse, and Henry Cowell (for whom he built the rhythmicon), and conductor Leopold Stokowski.

Glinsky's attention to historic detail vividly displays the arbitrary manner in which millions of Soviet citizens' lives were devastated by those in power. It's difficult to imagine how anyone could survive what Theremin was subjected to, let alone remain productive. Yet he outlived the political system that controlled—often destroyed—his work and kept him a second-class citizen. Glinsky portrays Theremin as a man determined to fulfill his creative urges despite continual setbacks. When Theremin emerges from the shadows in Moscow in an unexpected encounter with his old friend and theremin virtuoso Clara Rockmore, one cannot help but marvel at the melding of luck, cunning, and naïveté.

Theremin: Ether Music and Espionage is an important book for readers interested in the theremin or electronic music's history. The book is exhaustively researched and engagingly written, and it includes an insightful forward by Robert Moog. Glinsky's exceptional portrayal of Leon Theremin is more than a mere music-related biography. I recommend *Theremin: Ether Music and Espionage* to anyone who enjoys reading about 20th-century history or Soviet-American relations.

Overall EM Rating (1 through 5): 5

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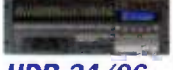

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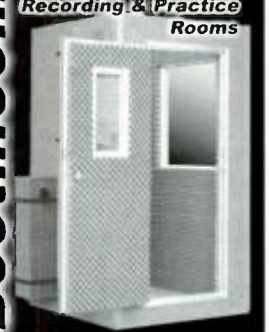

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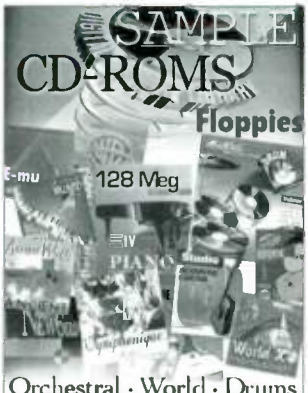

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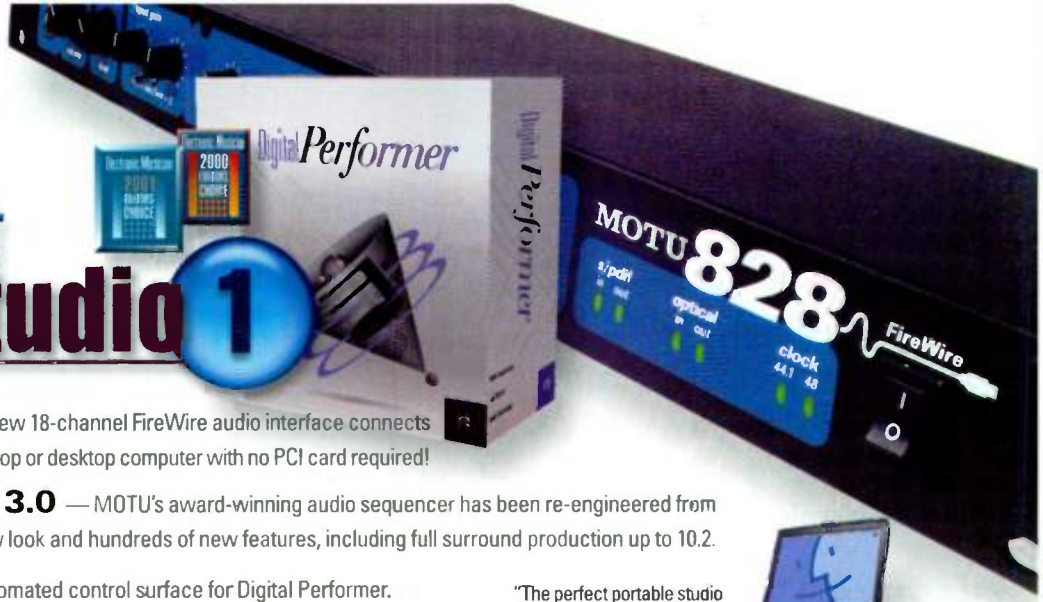

MOTU Dream Studio 1

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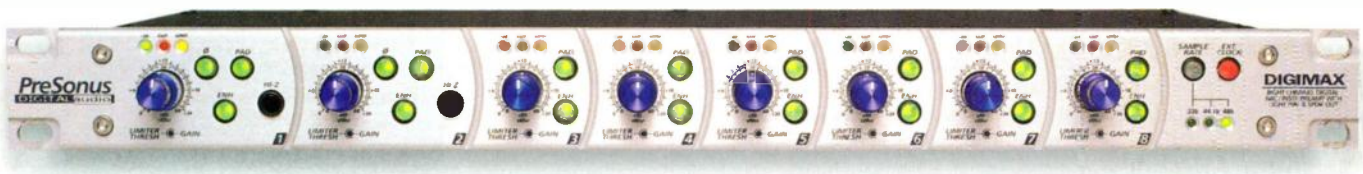

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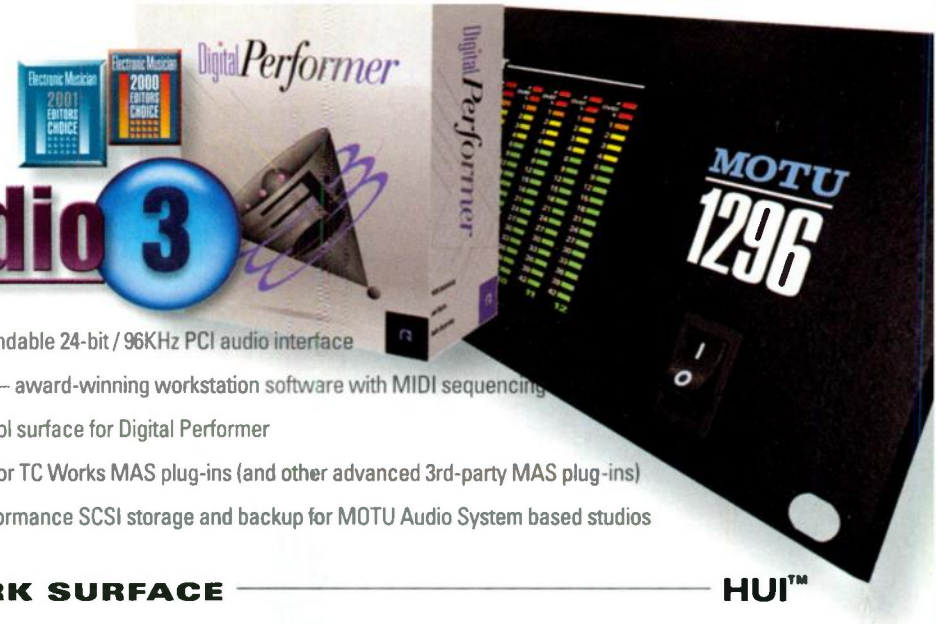

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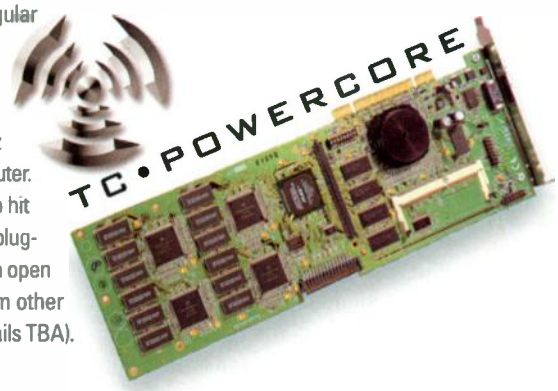

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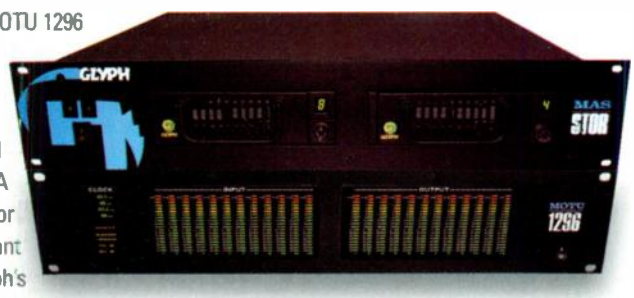

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By Larry the O

Stuck in the Middle

With Digidesign's mid-'90s introduction of its PostConform Pro Tools add-on (which played QuickTime movies in conjunction with a video accelerator card), affordable, all-digital post-production became a solid reality. Personal studios and multiroom facilities with many systems to buy took note. Being an audio-only system, Pro Tools developed and grew strong in the post market.

Flash-forward to today, when desktop computers play QuickTime just fine without add-on accelerators and all-digital production is old hat. The Pro Tools product line has mostly moved several thousand dollars upscale, leaving a gap in its wake. Those with the budget generally still opt for Pro Tools, but for everyone else, as my father never said to me, "It's not a bargain if you can't afford it." The alternatives for those left behind were unclear.

In the past few years, however, as ASIO has caught on, plenty of inexpensive I/O hardware for digital audio workstations (DAWs) has appeared. The burning question has been what software to use. With Pro Tools out of the picture (so to speak), the most powerful multitrack digital-audio editors for midrange users were digital-audio sequencers. They had integrated QuickTime or AVI playback but weren't really optimized for audio post other than music scoring: time was often expressed in bars and beats, busing and panning weren't conducive to surround work for film, spotting functions were weak, and so on. The products were certainly usable, but people got the feeling they were fighting the tide when they used a music-oriented sequencer package for post.

Now it seems the tide may be turning. A few years ago, Steinberg developed *Nuendo*, a program built from the ground up for post work, for Silicon Graphics (SGI) computers. More recently, the company worked up Windows and Mac versions and promoted them heavily at the recent winter NAMM show. Also shown at NAMM was Mark of the Unicorn's *Digital Performer* 3.0, which offers a host of new features targeted squarely at the surround and post-production market.

The designs of both *Nuendo* and *Digital Performer* show that their manufacturers have been listening to the market they are courting. The programs offer substantial surround-mixing capabilities, extensive fade control, shortcut definitions for every function, and more. Each program has its strengths and weaknesses. *Digital Performer* has the more versatile surround panning whereas *Nuendo* has a comprehensive, multilevel undo structure.

For those stuck in the middle, this is a nice trend, and it's likely to get better. I'd be surprised if Emagic wasn't cooking up its own package to get a chunk of the post market, and others will surely join the fray.

But taking on a new market means the manufacturers have to cover new territory. Although each of them has had customers in post-production, the increase in numbers is going to focus more attention on the post market's needs, such as extensive, direct user support. If you are a software developer and the work on James Cameron's next epic is held up by bugs in your software, immediate help is needed, and you will be made aware of that fact in no uncertain terms. It will be interesting to see who will manage the support challenge most effectively.

Post-production editors and mixers also demand stability and won't tolerate a new version ridden with bugs. Post-production houses will wait to purchase or upgrade until they are confident the program is stable enough to function in a high-pressure production environment. They also want their multitrack graphical user interface to display the program's audio portions and hide its music heritage (music scoring excepted, of course).

We can expect some bumps in the road; for a start, neither *Digital Performer* 3.0 nor *Nuendo* for the Mac are shipping as I write this. But it is gloriously clear that smack dab in the middle is, once again, a good place for post-production facilities to be. ❦

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