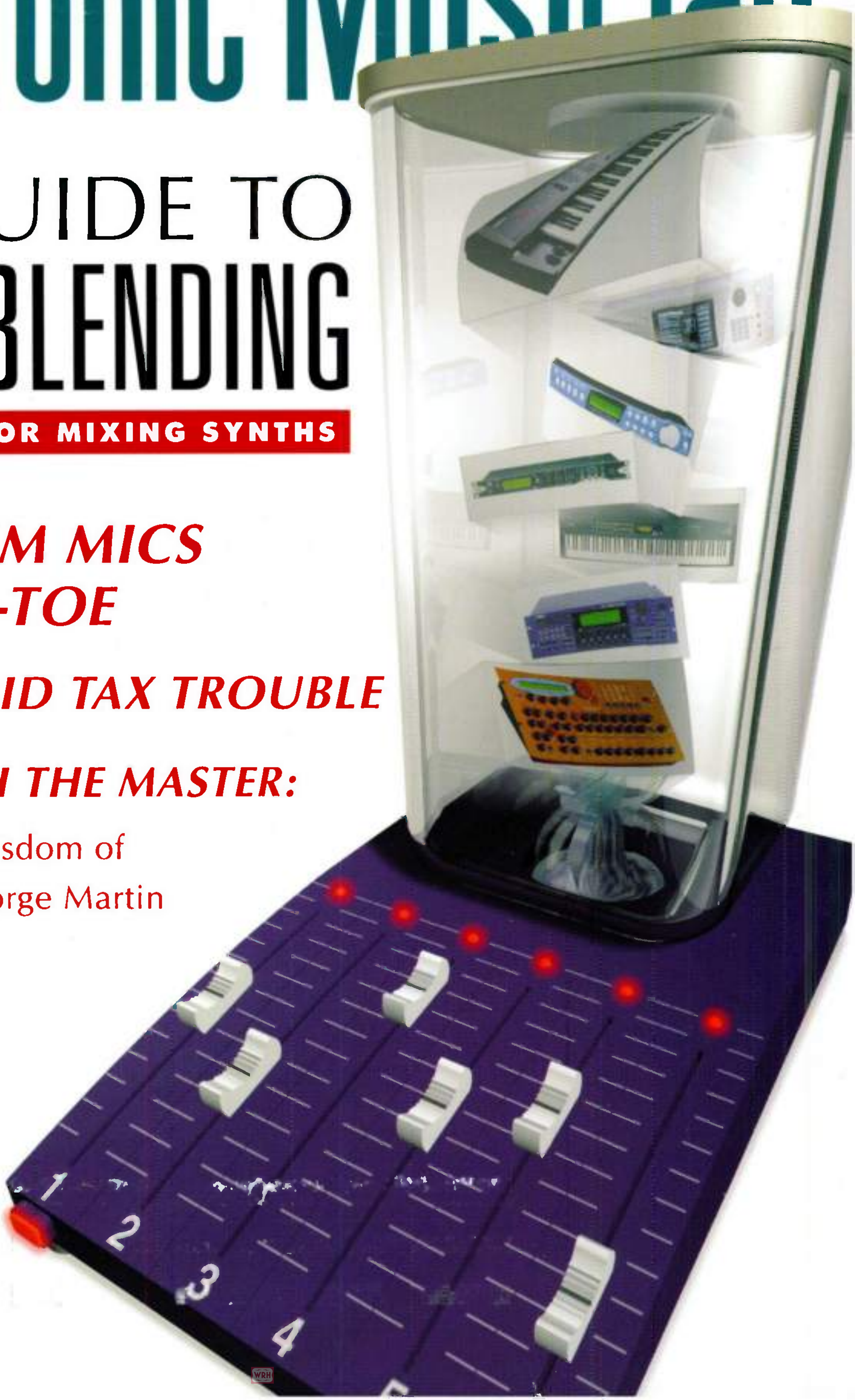
Reviews Bonanza: Yamaha DSP Factory, Cakewalk Pro Audio 8, Roland VS-840, and 9 more

# Electronic Musician®

February 1999



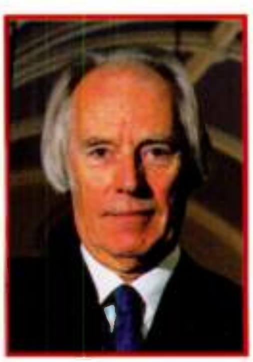

## EM'S GUIDE TO BETTER BLENDING

**ESSENTIAL TIPS FOR MIXING SYNTHS**

## *7 KICK-DRUM MICS GO TOE-TO-TOE*

## *HOW TO AVOID TAX TROUBLE*

## *A LESSON WITH THE MASTER:*



The Wisdom of Sir George Martin

A PRIMEDIA Intertec Publication

U.S. $4.95/Canada $5.95

02

0 71896 47104 5

# VETERAN ENGI

## THE GROUNDSWELL OF HR82

Everybody makes glowing claims about their monitor speakers. But only Mackie Designs' HR824 Active Near Field Monitors have gotten this amount of acclaim from credible outside sources so quickly.

Here are some verbatim comments gathered by our roving Mackie video crew on a recent visit to Nashville and Los Angeles (call us toll-free for a copy of the finished epic production), interspersed with recent review excerpts.

We know you're as serious about your creative product as these folks are. So why compromise with less than the best near field monitors? Visit your nearest Mackie dealer for a demo or call us toll-free for more info.

# HR824


**"Mackie asserts that the HR824s are 'smooth from 39 to 20kHz (±1.5dB)' and our tests corroborated the claim. This is no mean feat for monitors this size. The HR824s performed admirably, allowing us to distinguish very fine shades of tonal color and to establish subtle timbral and harmonic relationships between sounds. If you are in the market for a pair of compact active monitors and you are not afraid of the truth, do yourself a favor and give the Mackie Designs HR824s a critical listen."**

*"When I was tracking for Robert Redford's The Horse Whisperer, I put a lot of low end musical instruments onto the tape. When it came time to mix, no way could I have thrashed it out without the Mackie speakers. They really saved my life. My next job is in Calgary, Canada. I'm bringing four Mackies."* Brian Ahern, Engineering Legend

*"Very musical. Very accurate. We actually move them between our five rooms."* Glenn Meadows, TEC-nominated mastering engineer, Masterfonics


*"The most balanced pair of speakers I've ever had. I haven't heard anything better. The Mackies bring the full spectrum of sound into my room. They bring the full scope of the sound in an area that encompasses me AND my clients. You get subsonics — a terrific fullness of acoustic guitar, the lowest end of bass drum. When you have an upright bass you get low end that I normally don't get in a room like this."* Stephan Oberhoff, independent L.A. producer/ engineer/keyboardist

**"Performance, features and a cost-not-barred design at a retail price of $1500 a pair* make this product a very good value. In the words of one person involved in these listening tests, 'I have a feeling that [the HR824s] will become the NS-10 of the late '90s and beyond' ...ubiquitous."**

*"I love the [HR824's] bottom end — it sounds real. You don't have to compensate or guess. It's nice to FEEL a speaker. Producers also say they feel really good."* Stanley Smith, feature films soundtrack composer, co-producer of Jordan Hill



*suggested U.S. retail.

# EERS AND REVIEWERS CONFRONT REALITY.

## MONITOR RAVES BECOMES A TIDAL WAVE... AND OTHER MIXED METAPHORS.


**"HR824s give systems costing twice as much a run for their money in terms of sound quality... they deliver a solid low end that's surprising for their size and a flat transparent response across the spectrum."**

*"[HR824s] sound incredible — I was extremely surprised by the low end response. Clarity, detail and reproduction in reverb tails is real good."* Pat McMacon, Facility Director, Sony/Tree Studios

*"Very tight bass... clean mids... crystal pristine highs. There's a truth to them once you hear you can't go back."* Frank Serafine, Hollywood motion picture and television sound designer

*"Very clean. Spectacular. Very impressive. You can listen to them for a long period of time. They work in a lot of rooms. What we get on these speakers comes out when we take the tapes other places."* Milan Bogdan, General Manager, Emerald Studios (Billboard Magazine-rated "#1 Country Recording Studio")

*"On material I've mixed using another monitor brand, I'm now hearing things I missed. Imaging is wide and very even. The whole spectrum is equally represented. Great frequency response... midrange is smooth... no low end hypiness."* Bill Smith, Grammy-nominated recording engineer


**"Their treble output is detailed and extended yet very smooth. Words like 'open' and 'silky' come to mind... and these adjectives apply to a very wide sweet spot. The Mackies put out the kind of deep, warm bass normally considered the sole domain of huge drivers and subwoofers. I would consider these speakers a bargain at twice the price, but at list they are a steal."**

*"You can sit and listen to them all day long. You get exactly what you hear. They're totally natural — I can't say the word enough — NATURAL."* Lee Roy Parnell, Grammy-nominated singer/songwriter/producer

MACKIE.

MADE BY THE MACKIE DS IN WOODINVILLE, USA
800/898-3211
www.mackie.com

SEE US AT NAMM BOOTH #814

circle #502 on reader service card

# Never before has so much power been so affordable.

It's time for you to make the move to digital mixing. Analog mixers just don't offer the performance, features and sound quality required for today's digital recording and 5.1 surround mixing environments.

The Panasonic DA7 digital mixer gives you 16 mic channels with up to 24 channels of analog or digital I/O in ADAT, TDIF or AES format; 24-bit audio quality; moving faders and automation for less than you thought possible.

Quite a remarkable feat, but it's what you have come to expect from the company with more digital experience than anyone. In fact, Panasonic set the standard for digital recording years ago with the SV Series of DAT recorders.

So if you're still on the fence about digital mixing, we're certain the DA7 will convince you that now is the time to make your move.

*"The DA7 is extremely musical and easy to use in any recording environment. It offers a sound, warmth and punch that make it truly unique."*

Greg Ladanyi,
producer/engineer
Jackson Browne, Don Henley, The Violets, Fleetwood Mac, Jody Davidson

DA7 With products like the SX-1 and DX1000 (the DA7's big brothers), and the SV Series of DAT recorders, Panasonic is truly taking digital further. (Below - the DX1000 digital console.)

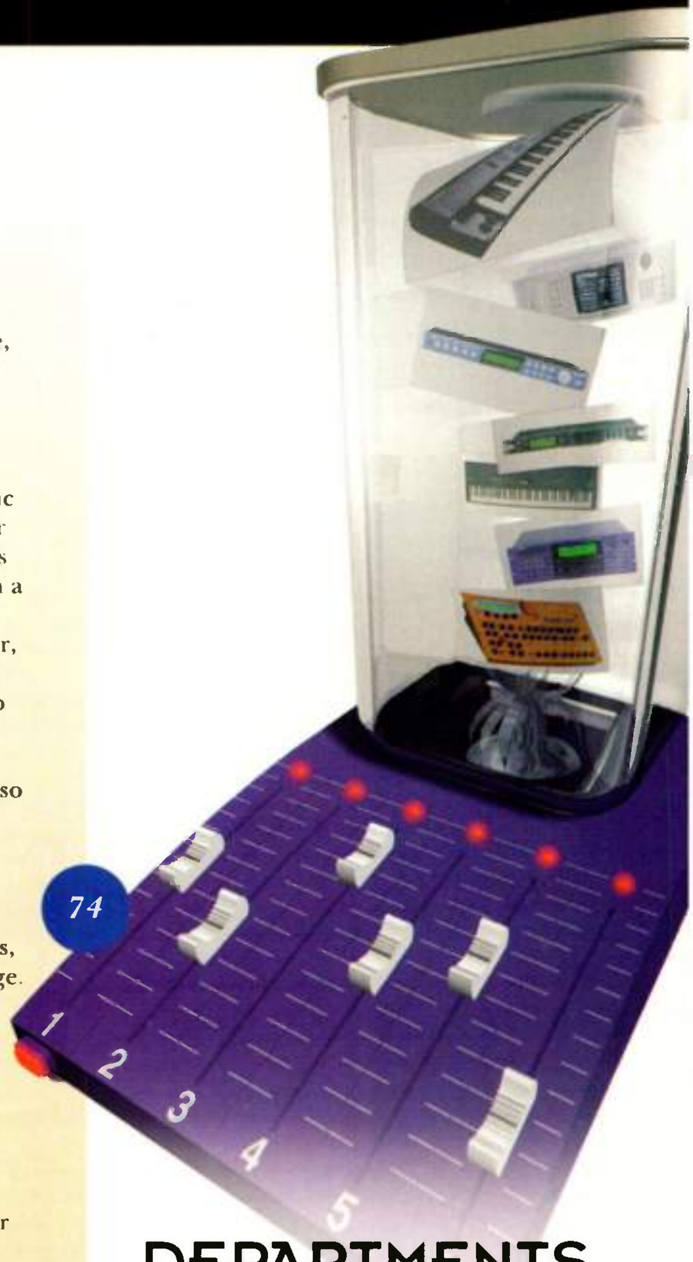
DA7 And many magazine editors have raved about the DA7's exceptional sonic quality, intuitive user interface, automation and affordability. (Some editors even bought the DA7 after the review!)


RIGHT PRODUCT.
RIGHT TIME.
RIGHT PRICE.
RIGHT NOW.
THE DA7
DIGITAL MIXER.
Only $170 a month!
THE ASSOCIATES
DA7 We've made it affordable. Now you can own a DA7 with our incredible 36 month, no money down, $170/month lease-to-own program.* See your dealer for details or call The Associates at 1-800-526-8638. Make your move to digital now.
* Program subject to customer credit approval and acceptance by Associates Commercial Corporation.
DA7 Don't just take our word for it. World-class producers, including Greg Ladanyi (pictured left) and Denny Purcell are using the DA7 for their high-profile projects.
"I could not have done my last four major 5.1 projects without the DA7."
Denny Purcell, producer/engineer
Reba McEntire, Colin Raye, Spaghetti Western, Joe Cocker
TAKING DIGITAL FURTHER.
Panasonic
Broadcast & Digital Systems Company
Call us to receive an informative brochure and reprints of the magazine reviews. Or simply visit our web site.
1-800-777-1146
www.panasonic.com/proaudio
WRH

# I N S I

## FEATURES






## DEPARTMENTS

# D E

Electronic Musician®
FEBRUARY 1999 VOL. 15, NO. 2

## COLUMNS



32

88

## REVIEWS



Cover: Illustration by Peter Neumann

# To Sir with Live

Got live if you want it.

ANTHONY PIDGEON

In the best **EM** tradition, we have something old and something new for you this month. In the first category, we have a wide-ranging interview with Sir George Martin, the legendary producer who is retiring after a long, hit-studded career (see "To Sir with Love" on p. 54). Most people associate Martin with the Beatles, of course, and some folks know his earlier work when he produced records by high-caliber comedians such as Peter Sellers. But did you know that Martin also produced significant albums by such artists as Seatrain, Jeff Beck, the Winter Consort, and Mahavishnu Orchestra? Indeed, his career was as broad as it was deep.

Given the scope of Martin's career, we wanted to offer the big picture of his approach to production rather than present just another Beatles retrospective. (Of course, we asked him some Beatles questions, too!) To accomplish this, we talked not only with Sir George but also with a wide variety of well-known musicians whose work he produced. The result is a unique perspective on Sir George Martin as he rides off, bathed in glory, into the sunset.

Now on to something new. **EM** readers are undoubtedly accustomed to our penchant for making occasional changes in the magazine. Like all dedicated musicians, we never want to stop improving. In that spirit, we have decided to upgrade our coverage of products, techniques, and technology for live performance. In the past, we handled these topics in a bimonthly *JAM* supplement; henceforth, they will be fully integrated into **EM**.

To begin with, we are introducing a new "Stage View" feature series, which will discuss live-performance technology, products, and applications. We're kicking off the series with "Playing to the Back of the House" (p. 100), Rudy Trubitt's story on how to use delay stacks in club P.A.s to achieve superior room coverage at moderate volumes.

This issue also introduces a new monthly column, "Performing Musician" (p. 144), which will focus on live performance issues and applications. Sometimes the column will feature familiar *JAM* authors and topics; for instance, former "Vox" columnist Joanna Cazden offers tips for singers in this month's column, and in our next issue, former "Tech" columnist Bean will take us on one of her popular high-tech explorations. As we roll along, we'll also introduce new authors and fresh perspectives.

To top it off, we're debuting a new section of "What's New" called "Performance Tools," which will present gear for the stage musician and sound engineer. This section of "What's New" will be treated like our "Sound Advice," "Get Smart," and other special "What's New" sections: it won't run every month, but you'll see it often throughout the year. That way, we can focus on performance tools that are really cool—the cream of the crop.

These changes represent our renewed commitment to covering topics of interest to musicians who perform live and to doing so at a higher level than ever before.


**Electronic Musician**®

**Editor** Steve Oppenheimer
**Managing Editor** Mary Cosola
**Associate Editors** Jeff Casey, Brian Knave, Dennis Miller, Gino Robair, David M. Rubin
**Copy Chief** Patricia Hammond
**Editorial Assistants** Carolyn Engelmann, Matt Gallagher, Rick Weldon
**Contributing Editors** Larry the O, George Petersen, Scott Wilkinson
**Art Director** Dmitry Panich
**Associate Art Directors** Tami Herrick-Needham, Laura Williams
**Graphic Artist** Steve Ramirez
**Informational Graphics** Chuck Dahmer

**Publisher** John Pledger
**Marketing Manager** Christen Pocock
**Eastern Advertising Manager** Joe Perry
**Midwest Advertising Associate** Julie Clark
**Northwest Advertising Associate** Joanne Zola
**Southwest Advertising Manager** Erika Lopez
**Sales Assistants** Alex Boyd, Mari Stancati
**Classifieds Advertising Manager** Robin Boyce-Trubitt
**Classifieds Sales Assistant** Jef Linson
**Classifieds Assistant** Mark Hopkins
**Sales Promotions Manager** Julie Shandrew
**Sales Promotions Coordinator** Daniela Barone
**Director of Operations and Manufacturing** Anne Letsch
**Production Director** Ellen Richman
**Advertising Production Coordinator** Amanda Weeden
**Production Assistant** Jeremy Nunes
**Computer Systems Coordinator** Mike Castelli
**Circulation Director** Philip Semler
**Circulation Associate** Heidi Eschweiler
**Circulation Assistant** Austin Malcomb
**Business Manager** Cindy Elwell
**Assistant to the Publisher** Monica Cromarty
**Human Resources/Facilities Assistant** Lauren Gerber
**Receptionist** Carrie Gebstadt

**National Editorial, Advertising, and Business Offices**
6400 Hollis Street #12, Emeryville, CA 94608
tel. (510) 653-3307; fax (510) 653-5142;
Web www.emusician.com

**Subscriptions, Customer Service, Back Issues**
PO Box 41525, Nashville, TN 37204
tel. (800) 843-4086 or (615) 377-3322; fax (615) 377-0525

**Intertec Publishing Corp.**
9800 Metcalf Ave., Overland Park, KS 66212

**PRIMEDIA Intertec**
Ray Maloney, President and CEO
Cameron Bishop, President, Communications & Entertainment Division
John Torrey, Group Publisher
Stepanie Hanaway, Division Director of Marketing

**PRIMEDIA Information Group**
Curtis Thompson, President

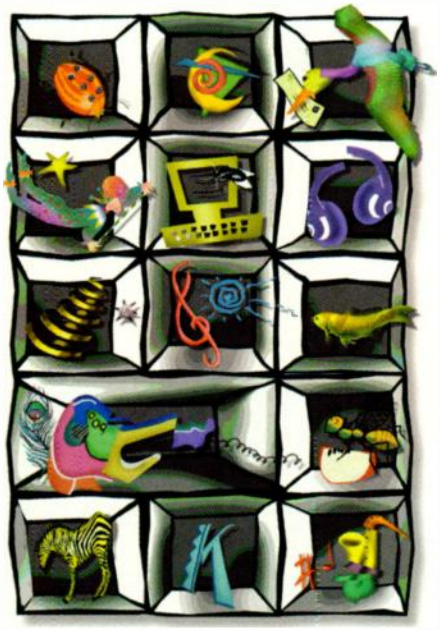
**PRIMEDIA Inc.**
William F. Reilly, Chairman and CEO
Charles McCurdy, President
Beverly C. Chell, Vice Chairman

**Electronic Musician** (ISSN: 0884-4720) is published monthly by **PRIMEDIA** Intertec 6400 Hollis St., #12, Emeryville, CA 94608. ©1999. This is Volume 15, Number 2, February 1999. One-year (12 issues) subscription is $36; outside the U.S. is $65. Periodical postage paid at Oakland, CA, and additional mailing offices.


POSTMASTER: Send address changes to Electronic Musician, PO Box 41525, Nashville, TN 37204. Editeur Responsable (Belgique): Christian Desmet, Vuurgatstraat 92, 3090 Overijse, Belgique. Canadian GST #129597951. Canada Post International Publications Mail Product (Canadian Distribution) Sales Agreement No. 0478741.



PRIMEDIA Intertec
Also publishers of *Mix*® magazine.
Printed in the USA.
BPA



## COPYRIGHTING WAV FILES

I am in the process of sending out my demo reel, and in addition to physical copies, I plan on sending it via the Internet as an MP3 (MPEG-1 Layer III) file. First of all, do you have any suggestions for better formats (that is, higher compression ratios or better sound quality)?

Second, how do I get copyright info onto a WAV file? When I click on a WAV file in Windows and look at the Properties/Details dialog box, there's a line that says "Copyright: No copyright information." I can't find any way of editing that line. I have tried typing text into every file- and track-related text box in *Cakewalk* before exporting the audio to WAV, but that doesn't seem to work. Leaving it as is, with the default "No copyright information," seems like an invitation to get ripped off.

**Jonathan Berridge**
**jman@skyinet.net**

*Jonathan—Distributing examples of your music over the Web as MP3 files is a great idea, because the MP3 codec (which is typically used to encode WAV and AIFF files) offers a reasonably high compression ratio while still delivering excellent fidelity. In fact, the MP3 format has become so popular that there are numerous Web sites devoted exclusively to the sale and distribution of MP3 files. For a closer look at how to create, handle, and distribute MP3 files, check out "Desktop Musician: The MPEG Audio Craze" in the January 1999 issue of* **EM**.

*Regarding your second question, I am not aware of any way to change a WAV file's copyright information at the system level in Windows. The information must be entered with the program that creates the file before the file is saved. Several audio-editing programs, such as Sonic Foundry's* Sound Forge *and Syntrillium Software's* Cool Edit Pro, *allow you to enter and save copyright information along with a file. Unfortunately, Cakewalk* Pro Audio *doesn't offer this feature.*

*Your compositions are automatically protected to some extent by the copyright laws as soon as the works are rendered in tangible form (written down or recorded on tape, CD, etc.). Nonetheless, I agree that leaving the "no copyright" label is just asking for trouble. Try importing your files into an audio editor and making the changes. On the other hand, if you subsequently import the files into an MP3 encoding utility, the copyright information may be lost during the conversion process anyway.—David R.*

## COMPRESSION SESSION

First of all, I'd like to say that I'm a longtime subscriber to your magazine, and I've been very happy with it.

I have a question about the article "Conquering Peaks" by Jeff Casey, in the December 1998 issue. I am a little confused on what he says about setting compressor release times according to tempo. Does this mean total beats per minute divided by release time in milliseconds? If so, what about live situations, in which you don't have the convenience of knowing exactly what the tempo is? You can only guess, and since compression is not exactly a thing that you can hear (other than making levels sound more consistent), setting the release time is a little tough. I know when my attack and release times don't jive, because that crazy phenomenon of pumping creeps in. This I can eliminate: I just set the attack and release fairly similarly at quick to moderate settings. This works for me, and I get by, although I don't think that I am benefiting as much as I could. I have read a lot of articles on compression, and there still is that inkling of mystery that lingers.

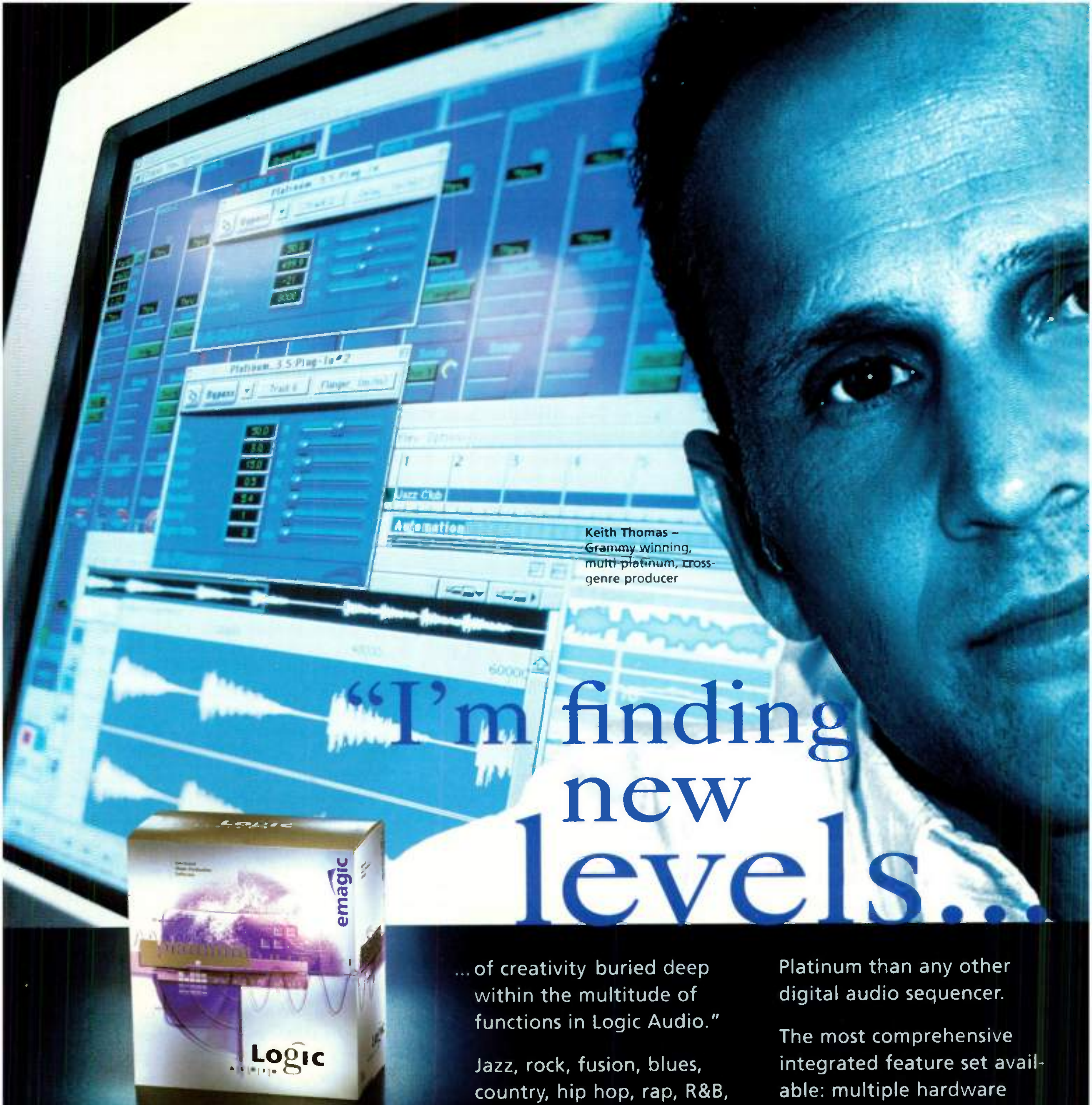
**Ken Pedraza**
**Hebron, IL**

*Ken—Actually, good compression is not something you can hear, but bad compression certainly is; if parameters are set improperly, you'll notice it. I agree that finding proper attack and release settings can be a tricky process. Unfortunately, this is not an exact science, and there is no formula that can be applied, so you're going to have to rely on your ears.*

*When I advise setting these parameters according to tempo, I'm referring to the tempo of the phrasing (as in the bass lines, vocal phrases, and snare drum hits) within the track being processed. In general, the faster the phrasing, the shorter the attack and release times should be. However, melodic instruments and percussive instruments are usually approached differently. For example, when you compress a bass guitar, your goal is usually to evoke sustain, so you'd probably want the release time to be as long as possible, extending just into the beginning of the subsequent phrase. On the other hand, the release on a vocal track is typically kept fairly tight, completing just after the end of each phrase. So there is a variable, but only within the fixed tempo of that track. When in doubt, I usually dial in a quick to moderate attack and*



BEN FISHMAN


Keith Thomas –
Grammy winning, multi-platinum, cross-genre producer
"I'm finding new levels...
emagic
Logic

*a moderate to slow release time and adjust both parameters from there.*

*Unless you're processing a sequenced track, there is going to be some movement in the song's tempo. In reality, though, this slight variation is not going to compromise the integrity of your compression settings. What can affect them is a total change in tempo, either of the track or of the song itself (like in a power ballad, for example). In these instances, your best bet is to use two separate compressors, which you can easily do if you're working with a DAW.*

*Of course, in live situations it's a little impractical to change compression settings for each new song. Even a band's regular sound engineer will have a tough time anticipating the exact tempo and phrasing of each player's performance. My advice is to make a compromise: dial in fairly quick attack times for all instruments; quick to moderate release times for vocals, percussion, and horns; and moderate to slow release times for guitars and synths.—Jeff C.*

## TIRED OLD BATTLE

I suppose that the letters titled Tempest in a Teapot and In the Other Corner, both in your November 1998 issue, serve to indicate that the platform warriors will keep doing battle right into the next century, never looking around to note that the war is over. By now it's pretty obvious that the Windows machine will engulf us all. It's also pretty obvious that the Mac will remain a vital niche technology for many years, of great importance to professionals in graphics and audio. So stop already.

I, for one, applaud **EM** for covering both camps equitably and fairly. I think we all ought to realize that the job of a periodical such as yours is to report on what's out there and what's happening right now, and you're doing it. Congratulations.

As for my good colleagues, the Windows zealots and Mac fanatics, remember: in polite company, never discuss politics, religion, or computer platforms.

**David Tcimpidis**
**DTcimpidis@aol.com**

*David—Well said; I agree with you wholeheartedly. We will continue to cover a balanced mix of Mac and Windows products. When products of interest to* **EM** *readers are developed for alternative operating systems, such as Be OS and Linux, we will cover those, too.—Steve O.*

## MIDI CONNECTION?

I have just started a home studio using a Pentium-based PC, a Korg keyboard, and a Tascam 414 Portastudio. The software I am running is PG Music *PowerTracks Pro Audio.* It is very similar to the setup described by Dennis Miller in the article "Build a Personal Studio on Any Budget," in the July 1998 issue of **EM.** I have a question: Dennis has a MIDI connection going from the Roland XP-10 directly to the Tascam 414—is this correct ? I was led to believe that MIDI didn't produce any sound—what gives ?

What should the connection be? I am connecting the sound card directly to the Tascam unit, but the sound from the AWE64 card output jack isn't great. What would you recommend or what am I doing wrong ?

**Mark Aboud**
**Brisbane, Australia**

*Mark—In answer to your first question, the connection between the Roland XP-10 and the Tascam 414 is an audio connection. We wanted to show that the XP-10's sound would be mixed with our other sound sources directly in the 414's mixer. MIDI messages are nothing more than instructions that tell a synthesizer or other MIDI-compatible device what to do. There's no reason to have a MIDI connection between these two devices, although some mixers and tape decks can be controlled via MIDI.*

*Next, I'm not sure what problem you are having with your AWE64 sound card. I've found it to be one of the best sound cards in its price range. Are you using its internal ROM sounds or the software-based sound set that is installed on your computer when you added the card? Also, check the levels in the AWE64's internal mixer, which you should see represented by a yellow speaker icon in your taskbar. Perhaps you don't have the output level set very high and, as a result, are raising your mixer's faders higher than needed. There are numerous things that can affect what you hear in your mixer, but without a better description of your current configuration, I don't know what to suggest next.*

*I recommend that you carefully read through the setup instructions that came with your card, that you check for the latest drivers at Creative's Web site, and that you post a question or two in one of the audio newsgroups if you're still unhappy with the card. I'd be surprised if there weren't some simple tweaks you could make to get satisfactory results.—Dennis M.*

## SYQUEST SHUTS DOWN

I'm writing to warn **EM** readers not to buy SyQuest hard disks. Although SyQuest drives are still being sold, SyQuest has gone out of business, and the drives that are available are unreliable.

I bought a SyQuest SparQ removable 1 GB hard disk in mid-August for audio recording. It failed at the end of October. When I would insert a disk, the drive would start spinning up, spin back down, and eject the disk. The tech-support operator gave me the impression that my problem was a common one.

Shortly afterward, I read an *InfoWorld* article titled "SyQuest Technology Shuts Its Door for Good" (*InfoWorld,* November 9, 1998, p. 26). According to the article, SyQuest is considering filing for Chapter 11 bankruptcy but claims that it will maintain a limited staff to support its customers. *InfoWorld* found that the SyQuest Web site had been shut down and was unable to contact the company by telephone. My attempts to contact SyQuest after reading the article ended in failure.

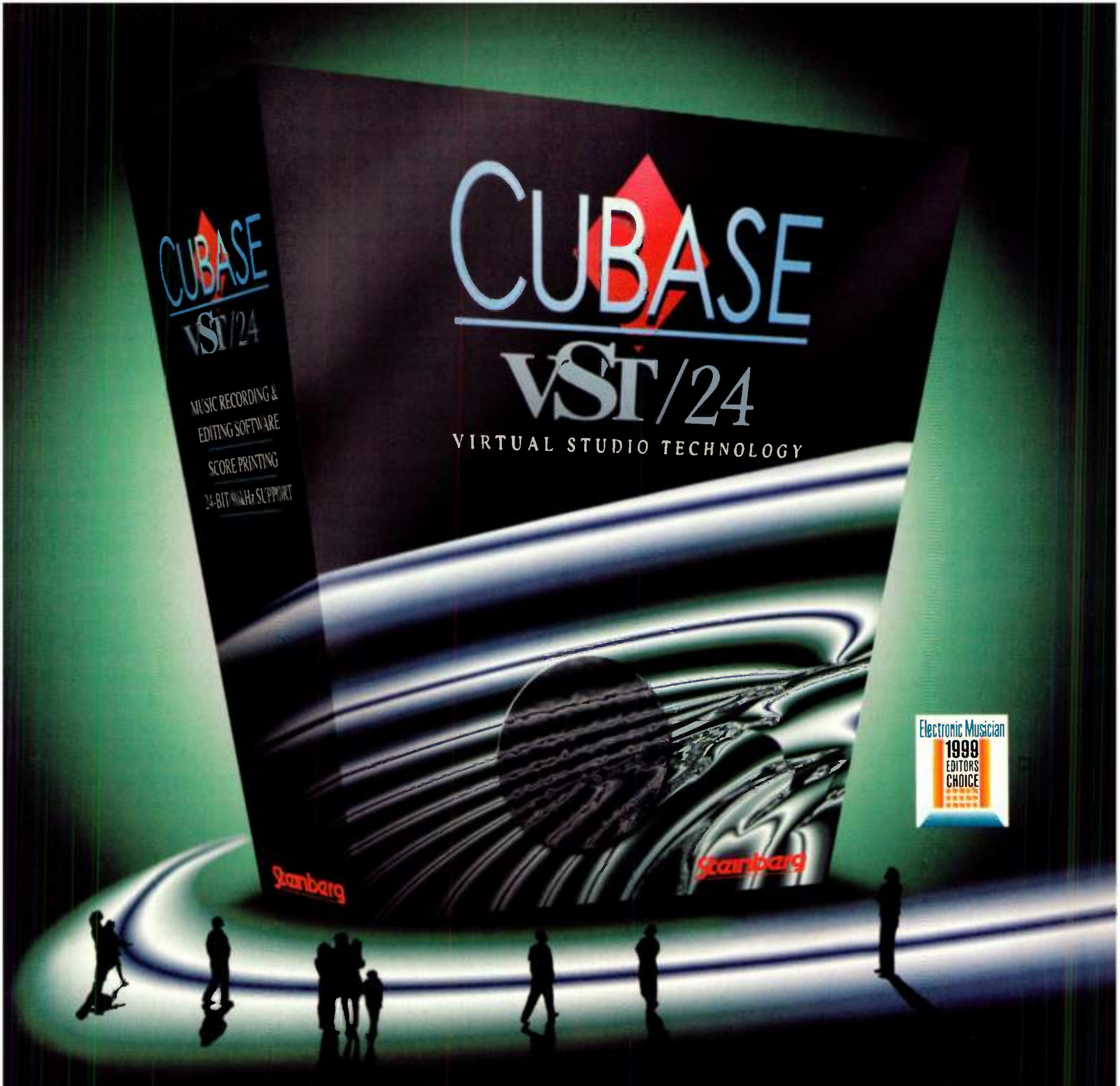
So I'm stuck with a $200 disk drive and another $100 in useless removable disks. Fortunately, I hadn't used the SyQuest to store any irreplaceable audio files. I hope this information saves other **EM** readers from encountering similar problems.

**Jim Olinger**
**jolinger@onr.com**

*Jim—Although we can't speak to the unreliability of the SparQ, we can verify that SyQuest has, in fact, filed for Chapter 11. Buyers should be cautious when purchasing new or used SyQuest equipment, because customer support can no longer be guaranteed. Although SyQuest-format cartridges are still available from third parties, long-term media availability is uncertain, as well.—Carolyn E.*

**WE WELCOME YOUR FEEDBACK.**

*Address correspondence and e-mail to* **"Letters,"** Electronic Musician, *6400 Hollis Street, Suite 12, Emeryville, CA 94608 or emeditorial@intertec.com. Published letters may be edited for space and clarity.*




# Making your music better, bit by bit.

Introducing Cubase VST/24* for Windows and Mac OS platforms. Same user-friendly layout and powerful plug-in architecture as before, but now with the uncompromising audio quality of 24-bit. Top it off with the flexible integration of hardware from manufacturers like Lexicon, Apogee, Korg, Yamaha and Event, and it's easy to see why Steinberg towers above the rest. New Cubase VST/24 from Steinberg. The future's sounding better all the time.

21354 Nordhoff Street, Suite 110, Chatsworth, CA 91311 (818) 993-4161 Fax: (818) 701-7452 Fax On Demand: (800) 888-7510
Steinberg Canada: (416) 789-7100 Fax: (416) 789-1667
www.us.steinberg.net

**Steinberg**
**NORTH AMERICA**


* Version 3.6 for Windows and Version 4.0 for Mac OS.

**circle #506 on reader service card**

# The Best Just Got

When you're looking for the top quality professional-standard mixer with plenty of inputs, road-worthy construction, versatile auxes, terrific EQ, and on-board effects, the new WZ16:2DX is it.

Building on the unprecedented success of the original WZ16:2, the new DX model has the winning features for the broadest range of applications from sound reinforcement to digital recording, plus a host of new assets, including onboard dual effects processor, all-metal connectors, larger channel-on buttons and LEDs and more.

*Wide-gain front end matches any source on any mono channel*

*Low-cut filters on all channels cuts mic thumps & stage rumble*

*4-band 2-sweep EQ for excellent sonic control*

*MSP signal path for sonic transparency (distortion less than 0.009%, better than 105dB dynamic range and an EIN of 129dBu)*

*6 Aux sends for lots of fx and outputs pre or post fade*

*Channel-on switches and LEDs show you what's going on - no pun intended!*

*Channel peak LEDs indicate potential clipping*

*Inserts on all channels for compressors, gates dedicated fx etc*

*16 XLR mic/line inputs - 20 incl stereos*

*100 mm faders, full length for smooth precision control*


ALLEN & HEATH

**ALLEN & HEATH USA** • 8760 South Sandy Parkway • Sandy, Utah 84070

# Better! MixWizard series

## WITH DIGITAL EFFECTS ON-BOARD

The WZ16:2DX incorporates the unique Allen & Heath QCC (quick change connector) system allowing the input connector block to be swung up for desktop or flightcase use, or down for rack mount operation.

Its MSP (minimum signal path) architecture has been designed by Allen & Heath engineers to assure sonic transparency from this new, professional mixer.

*If you're in the market for a new mixer, do it right... See the new professional standard at your Allen & Heath dealer.*


*Phantom power for inputs, no need for remote +48V supplies*

*Balanced XLR outputs for long cable runs without noise*

*12-segment 3-color meters, for clear, precise level control*

*Internal power supply means no fussy external power supply*

*Stereo or dual mono effects allows selection from 32 on-board effects*


*Stereo/Mono aux monitoring easy checks, very versatile monitors*

*Extra AB output provides independent line feeds for recording etc*

*QCC connector panel for deck, flight case or rack mounT.*

*16 all-metal XLR and 1/4" connectors on inputs means no more hunting for adapters*

*Edit FX presets via MIDI as well as provide program change information*


**Tel: (801) 568 7660** • Fax: **(801)** 568 7662 • http://www.allen-heath.com

circle #507 on reader service card

# WHAT'S NEW

By Carolyn Engelmann and Rick Weldon

## ▲ STEINBERG ORANGE VOCODER

Up until a few years ago, vocoders appeared to have gone out of style. Now they're back, both as hardware effects processors and as software plug-ins. In the latter category, Steinberg has introduced *Orange Vocoder* (Mac VST; $199) a vintage-vocoder simulator.

*Orange Vocoder* provides 24-band analog-vocoder simulation with 64-bit internal processing. It can use its built-in oscillator, a VST audio track, or any audio input source as the carrier signal and a VST audio track as the modulator. A Channel Flip button allows you to instantly exchange the carrier and modulator signals.

An 8-voice analog-emulating synthesizer is included in *Orange Vocoder,* with two oscillators per voice, ten basic waveforms and seven sampled sounds, voice detuning, and pitch LFO. The synth also includes a ring modulator and a 4-pole lowpass filter with cutoff and resonance. Rounding out this plug-in's features are the Freeform EQ and the postvocoder Filterbank Reverb. Steinberg North America; tel. (818) 993-4161; fax (818) 701-7452; faxback (800) 888-7510; e-mail info@steinberg-na.com; Web www.us.steinberg.net.

*Circle #401 on Reader Service Card*

## ▶ YAMAHA D24

Yamaha's 3U rack-mount D24 modular magneto-optical recorder ($3,499) can record eight tracks of 24-bit, 48 kHz digital audio or four tracks of 24-bit, 96 kHz digital audio onto 320 or 640 MB, 3.5-inch magneto-optical (MO) removable cartridges. Sixteen- and 20-bit sample resolutions can also be used, as can 44.1 and 48 kHz sampling rates, and sample rate and resolution can be set individually for each song on a disk.

Using a 640 MB MO disk, you can record up to 15 minutes of 8-track, 16-bit, 44.1 kHz audio; at 24-bit, 96 kHz, you can record up to 9 minutes of 4-track audio. Each track includes eight virtual tracks, and on a given MO disk, you can record up to 512 songs, each with up to 99 locate points. A SCSI-2 connector allows you to record to a hard drive.

The D24 can sync to MTC, SMPTE time code (bit-accurate), word clock, and video sync. It can be controlled via MIDI Machine Control and RS422 9-pin protocols or via an optional remote control. Using the unit's 15-pin D-sub remote port, up to eight D24s can be synched to create a 64-track system.

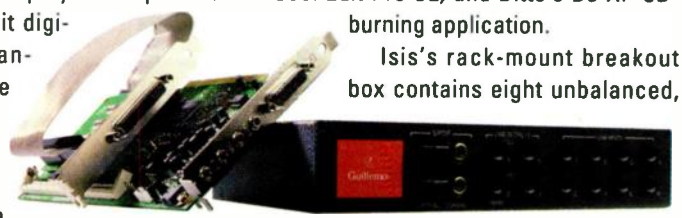
Editing features include Copy, Copy Insert, Move Insert, Move Replace, Erase, Delete, Loop, and Merge. Audio is stored in a nondestructive edit buffer, and the D24 allows for five levels of undo. Time compression and expansion are offered (two tracks at a time, from 50% to 200%, without pitch change). You also can change the pitch by ±6% (two tracks at a time) without changing tempo or duration. There is a Jog/Shuttle wheel for easy location of recorded material.

The D24 uses the same mini-YGDAI I/O cards as the company's 01V digital mixer. You can purchase eight channels of analog I/O or digital I/O in ADAT, TDIF, or AES/EBU format. Yamaha Corporation of America; tel. (714) 522-9011; fax (714) 739-2680; e-mail info@yamaha.com; Web www.yamaha.com.

*Circle #402 on Reader Service Card*

## ▼ GUILLEMOT MAXI STUDIO ISIS

Guillemot has released the Maxi Studio Isis ($399.99), a full-duplex PCI sound card and audio interface package for the PC that can record up to eight channels and play back up to four channels of 20-bit digital audio simultaneously at sample rates of 32, 44.1, and 48 kHz. The Isis package includes a 64-note polyphonic, General MIDI- and GS-compatible software synth and a sampler with two LFOs, three envelopes, and an adjustable resonant filter with cutoff slopes of up to 24 dB/octave. Guillemot has worked with Roland Corporation to put together the 4 MB worth of sample sounds included with Isis, and the sample memory is expandable to 36 MB using standard SIMM chips.



The card's onboard DSP is provided by a RISC-based Dream chip, freeing your CPU from the task of audio processing. Bundled software demo programs include Steinberg's *Cubase, ReBirth,* and *WaveLab;* Cakewalk's *Pro Audio* 8, Syntrillium's *Cool Edit Pro SE,* and Ditto's *DJ XP* CD-burning application.

Isis's rack-mount breakout box contains eight unbalanced, ¼-inch, line-level analog inputs; four unbalanced, ¼-inch, line-level outputs; and both optical and coax S/PDIF I/O. On the PCI card are a digital game port/MIDI port, stereo line-level input and output on ⅛-inch TRS connectors, and a microphone input on an ⅛-inch jack. Guillemot; tel. (514) 490-2161; fax (514) 279-4954; e-mail support@guillemot.com; Web www.guillemot.com.

*Circle #403 on Reader Service Card*

## ▶ E-MU PROTEUS 2000


E-mu's Proteus 2000 sound module ($995) combines the features of its E4 sampler line with advanced sound-generation capability in a 1U rack-mount unit. The Proteus 2000 is 128-voice polyphonic and 32-part multitimbral and has two independent MIDI In jacks.

The unit comes with 32 MB of RAM, which can be increased to 128 MB using 32 MB SIMMs. E-mu supplies a 32 MB sound set to get you started. A Sound Navigator feature eases sound management by allowing you to organize presets by type.

Each of the four front-panel parameter knobs controls three layers of parameters, giving you real-time control of up to 12 parameters. Sound-sculpting features include 6-pole dynamic filters and a 24-bit, dual-stereo effects processor that provides reverb, delay, flange, distortion, and so on. Users can save complete scenes, including preset numbers and volume and pan settings in Multi-mode Setups. Another feature automatically generates a "riff" you can run while auditioning sounds, freeing you from your keyboard while you tweak the Proteus 2000.

The unit has six ¼-inch analog outputs and an S/PDIF output pair. On its Web site, E-mu will provide operating system updates that can be uploaded to the Proteus 2000 via MIDI. E-mu Systems; tel. (831) 438-1921; fax (831) 438-8612; e-mail info@emu.com; Web www.emu.com.

*Circle #404 on Reader Service Card*

## ▶ GADGET LABS WAVE/8•24

The Gadget Labs Wave/8•24 (Win, $499.95; Mac, $599.95) 24-bit, 8-channel digital recording interface includes a PCI card; an outboard, 8-channel analog I/O unit; and a software driver. The Windows 95/98/NT drivers are available now; Mac ASIO support is expected by the end of March.

The 24-bit, 128x oversampling A/D/A converters are on the PCI card and support sample rates of 11.025, 16, 22.05, 24, 32, 44.1, and 48 kHz. Output can be 8-, 16-, or 24-bit. The card includes Gadget Labs' SoundCache chip, which handles audio formatting and routing.

Channels 1 and 2 have XLR mic-level inputs, XLR outputs (for use with a power amp), and ¼-inch inputs and outputs. The other six channels have only ¼-inch I/O connections. All channels can accept balanced or unbalanced signals and can operate at +4 dBu or -10 dBV. A 24-bit S/PDIF digital I/O daughtercard is available ($129.95). MIDI In and Out jacks are provided on the I/O unit's back, along with connections for sync cables. You can link up to three Wave/8•24 interfaces. The analog I/O interface connects to the PCI card with a 6-foot cable, and an optional 23-foot cable is also available ($24.95).

The Windows version requires at least a Pentium 166 with a PCI bus and 32 MB RAM and works with a variety of multitrack recording programs. Gadget Labs includes Syntrillium's *Cool Edit Pro SE* in the software package for basic multitrack capability, and demo versions of the company's own *WaveWARM* plug-in and *WaveZIP* file compressor are also part of the deal. The Macintosh version will run on a G3 200 or faster processor with Mac OS 7 or higher and at least 32 MB RAM. The Macintosh version will support Sound Manager and will include bundled software, but details have not yet been announced. Gadget Labs; tel. (503) 827-7371; fax (404) 685-0922; e-mail info@gadgetlabs.com; Web www.gadgetlabs.com.

*Circle #405 on Reader Service Card*

## ▶ GLYPH TECHNOLOGIES DIGDAT

The DigDAT ($995, tabletop unit; $1,150, rack-mount model) from Glyph Technologies is a 4 mm DAT-tape backup device that offers up to 6.8 GB of uncompressed digital storage (using 124m DAT tape) with a Centronics SCSI-2 interface.

Designed specifically for audio/video use, the DigDAT can stream up to two hours of MPEG-2 video (at 800 x 600 pixels and 30 frames per second) or up to 12 hours of MPEG-1 video to your computer in real time. It has a data transfer rate of 902 KB/second and an archiving rate of 521 MB/minute (typical). The DigDAT's SCSI burst transfer speeds are rated by Glyph at up to 10 MB/second (synchronous) and 7.5 MB/second (asynchronous). Access time for a 120m tape is rated at less than 40 seconds. The machine also supports DDS-1, DDS-2, and DVDS tapes.

The DigDAT is compatible with E-mu's Darwin; the Akai DR-8 and DR-16; Macs running Optima's *DeskTape*, Dantz's *Retrospect*, or Grey Matter's *Mezzo;* and PC workstations using Indigita *Tape Trax*. Included with the unit are a double-shielded, twisted pair, SCSI interface cable; external terminator; IEC power cable; and a blank tape. Glyph offers a one-year limited warranty on the device. Glyph Technologies; tel. (800) 335-0345 or (607) 275-0345; fax (607) 275-9464; e-mail info@glyphtech.com; Web www.glyphtech.com.


*Circle #406 on Reader Service Card*

# MICROPHONE CABINET ▲ ▲ ▲ ▲

## ▲ RØDE

The Røde NTV tube condenser microphone ($1,195) features a cardioid polar pattern and uses a 1-inch diameter, 6-micron thick, 24-karat gold–sputtered Mylar diaphragm. The microphone is externally biased, meaning there is a circuit that always applies a charge to the diaphragm. On the NTV, the electrical contact is made by a series of wires attached to the edge of the diaphragm. According to Røde, this design allows the diaphragm to move more freely, resulting in a better low-end response.

The NTV uses an ECC 81 twin-triode tube. The mic's preamp circuit is designed to use a minimal signal path, and a custom Jensen transformer is used inside the microphone. A 30-foot, double-shielded cable outfitted with oxygen-free copper and gold-plated contacts connects the NTV to its power-supply box.

The body of the mic and its dual-mesh grille are made from machined stainless steel. Also included with the NTV are Røde's SM2 suspension shock mount and M2 stand-mount adapter, and an aluminum flight case.

Røde rates the microphone's frequency response at 20 Hz to 20 kHz, self-noise at <19 dB (A weighted), and maximum sound-pressure level at 130 dB. Røde/Event Electronics (distributor); tel. (805) 566-7777; fax (805) 566-7771; e-mail info@event1.com; Web www.event1.com.

*Circle #407 on Reader Service Card*

## ▼ AUDIO-TECHNICA

The Audio-Technica AT4060 ($1,695) cardioid vacuum tube mic is designed to withstand high sound pressure levels (up to 149 dB SPL at 1 kHz at 0.5% THD) and offers an extended low-frequency response.

The AT4060's tube is hand-selected, aged, and individually tested, and two 2-micron, gold-vaporized diaphragms are used. These are externally biased and powered by the included AT8560 half-rackspace power supply. The power supply can be run off 120 or 230 VAC outlets and is outfitted with a ground-lift switch. Internal floating components and the included AT8447 shock mount provide isolation from vibration. A carrying case is also included.

Audio-Technica rates the AT4060's frequency response at 20 Hz to 20 kHz, self-noise at 19 dB (A weighted), dynamic range at 131 dB, and signal-to-noise ratio at 75 dB (1 kHz at 1 Pa). Audio-Technica; tel. (330) 686-2600; fax (330) 686-0719; e-mail pro@atus.com; Web www.audio-technica.com.

*Circle #408 on Reader Service Card*

## ▼ ROYER

Royer has designed its R-121 velocity-type ribbon mic ($995) to take advantage of developments in materials and mechanical construction. The company claims the mic is not affected by heat or humidity and delivers a smooth frequency response, free from high-frequency phase distortion. The R-121 has a figure-8 polar pattern, which Royer suggests makes the mic a good choice for room-miking, such as for orchestral and choral recordings.


The microphone's ribbon transducer assembly incorporates neodymium magnets in a flux-frame circuit, forming the magnetic field around the 2.5-micron thick corrugated aluminum ribbon. Royer claims that this design delivers greater sensitivity while reducing stray magnetic radiation. The mic's casing is made from burnished satin nickel.

The R-121 comes in a wooden case, and Royer offers an optional suspension shock mount ($72) and foam and screen pop filters ($20 and $47.50, respectively). Except for the ribbon element, the mic is guaranteed for life to the original owner. Royer offers the first re-ribboning free of charge and states that, with normal studio use, the microphone may never need to be re-ribboned. The company rates the R-121's frequency response at 30 Hz to 15 kHz (± 2 dB) and maximum SPL at 130 dB. Royer Labs; tel. (818) 760-8472; fax (818) 760-8864; e-mail info@royerlabs.com; Web royerlabs.com.

*Circle #409 on Reader Service Card*

# Ultra Power

## Introducing the NEW Emulator® 4 Ultra Sampling System.

**Faster, better, and more upgradeable. The most-advanced sampler in the world.**

E 4XT ULTRA

Emulator 4 Ultra Sound Library
- Producer Series
- Classics I and II Series
- Formula 4000 Series
- The E4 Definitive Series NEW
- Over 25 Titles

**ULTRA FAST** – E4 Ultra's new 32-bit processor gives you ultra fast MIDI response time, faster SCSI, and SMIDI operation, plus quick access to your most- used functions.

**ULTRA UPGRADES** – E4 Ultra offers new affordable ways to upgrade your sampling system, including the new ADAT 16 output/8 input card, new ROM and Flash memory boards, and the up-and-coming, revolutionary 32-bit R-Chip, 32 channel FX card from E-MU.

**ULTRA SOFTWARE** – E4 Ultra is complete with the EOS version 4.0 operating system, giving you new ways to interact with your computer. Also available for E4 Ultra is EOS Link, new software which allows you to control your Emulator from your Mac/PC computer – Load, Save, Sample, Edit, Browse – everything is a mouse-click away.

**ULTRA FIDELITY** – Simple. Our new samplers sound the best.

**ULTRA CHOICE** – Choose from three different E4 Ultra models, and get the one that best suits your style and budget: E4XT Ultra, E-Synth Ultra, and E6400 Ultra.

**ULTRA HIP** –E4 Ultra introduces a cool new real-time DSP tool that analyzes your drum loops and phrases, automatically finding BPM and the perfect loop for your sample. Now, change your loop from 4/4 to 7/8, swing your four on the floor to a funky Hip-Hop groove, and move beats in and out or switch their order-- all in real-time. We call this "Beat-Munging" and the E4 Ultra is the only sampler that can do this.

### E4XT ULTRA STANDARD FEATURES

- 64 MB RAM (expandable to 128)
- Word Clock I/O
- 32 MIDI Channels
- 8 Balanced Outputs (expandable to 16)
- 128-Voice Polyphony
- 3.2 GB Hard Drive
- Dual 24-Bit Stereo Effects
- AES/EBU Digital I/O
- 9 CD-ROMs (over 2GB of sounds)

**Upgrade Your Existing Sampler to Ultra Status** –
Contact E-MU's customer service department at 831-438-1921, or visit us at www.emu.com, for more details.

**E-MU SYSTEMS**

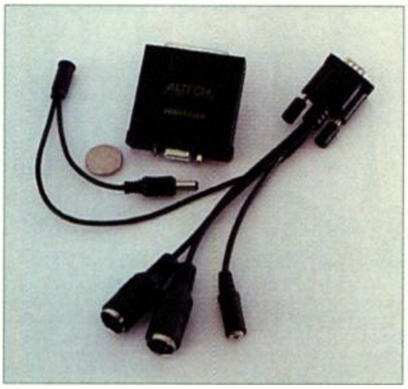
*West Coast Office*
1600 Green Hills Road
PO Box 660015
Scotts Valley, CA 95067-0015
831-438-1921

*East Coast Office*
155 Great Valley Parkway
PO Box 3035
Malvern, PA 19355-0735
610-647-3930



SEE US AT NAMM BOOTH #W1233

circle #508 on reader service card



## ▲ ALTECH SYSTEMS WAVEDOCK

Designed for use with laptop PCs and able to fit into a shirt pocket, Altech Systems' WAVEdock ($199.95) comprises a tiny GM-compatible wavetable synthesizer and parallel-port MIDI interface. The MIDI interface features two separate sets of ports, each with a MIDI In and Out connector.

When the unit is in PC-Docked mode, one pair of ports works with the internal wavetable synth and the other pair is a regular external MIDI interface. In Stand-Alone mode, WAVEdock works without a computer. Here, only one set of MIDI ports is active: data appearing at the MIDI In is fed to the internal synth and also is rerouted to the MIDI Out (which acts, in effect, like a MIDI Thru).

If you use it only as a MIDI interface, the WAVEdock does not need power when docked to a computer. If you want to use the synth, however, you have to connect the unit to the laptop's power supply via a T-style splitter connector (included). In Stand-Alone mode, you need an external supply. An optional battery pack provides four to five hours of use with two D cells.

The 32-note polyphonic, 16-part multitimbral synth offers 128 melodic instrument sounds and 47 percussion sounds. Independent digital reverb and chorus levels can be set for each MIDI channel on the internal synth. The synth outputs 16-bit, 44.1 kHz stereo sound via two 3.5 mm connectors.

The WAVEdock requires an 80386 or better PC with a clock speed of at least 25 MHz. Altech Systems; tel. (888) LUV-MIDI or (818) 709-4732; fax (818) 709-4039; e-mail pmontes@altechsystems.com; Web www.altechsystems.com.

*Circle #410 on Reader Service Card*

## ▼ TAYTRIX OVAL WORKSTATION

The two-tiered, geometrically inspired Oval Workstation ($3,050) is designed to be as stylish and safe (no sharp corners) as it is ergonomic. The Oval Workstation is available in two sizes, each with a countertop, adjustable raised meter-bridge shelf, and two swivel-mounted speaker platforms.

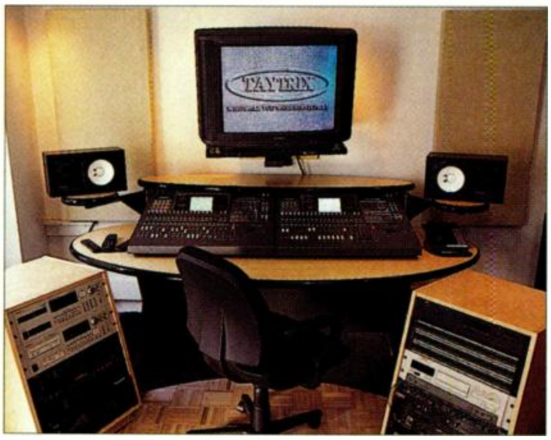
The standard model has a 6.5 x 3-foot oval tabletop and a 5 x 1.3-foot meter-bridge shelf. The total depth of the workstation is 42 inches. The tabletop of the Mini-O model measures 5 x 2.6 feet, with a 4 x 1.2-foot meter-bridge shelf and an overall depth of 38.75 inches.

Each model has a pair of speaker shelves that swivel 180 degrees and can handle up to 50 pounds each. The speaker shelves are affixed to the meter-bridge shelf, which can be vertically adjusted so that it sits from 6 to 18 inches above the primary oval.

Decorative black T-mold edging borders the furniture-grade plywood surfaces, which are available in your choice of maple or cherry to complement your studio decor. A steel frame with an integrated modesty panel supports the workstation, and an optional underdesk keyboard drawer is available.

The Oval workstation can also be purchased as a package with two RA12 angled 12-space racks for $3,400. Taytrix; tel. (201) 222-2826; fax (201) 222-5457; e-mail feedback@taytrix.com; Web www.taytrix.com.

*Circle #411 on Reader Service Card*




## ▼ APOGEE ROSETTA

The engineers at Apogee set out to produce a high-quality analog-to-digital converter that would be affordable to personal-studio owners, and the result is the Rosetta AD, the first unit in the company's new Solution series. This 1U rack-mount converter box is available in two models: one that supports 44.1 kHz and 48 kHz sample rates ($1,295) and one that handles sample rates from 44.1 kHz up to 96 kHz ($1,995).

The Rosetta AD supports 24-bit conversion and can generate 16- or 20-bit output using Apogee's UV22 high-resolution word-length reduction process. The two XLR analog inputs can accept either balanced or unbalanced, +4 dBu or -10 dBV input signals. Output can be sent in TDIF, S/PDIF coaxial, S/PDIF optical, or ADAT format. A pair of AES/EBU interfaces is also included for simultaneous output to two devices.

The simple front-panel design has dedicated controls for sample rate, resolution, and output selections. Another button activates the Soft Limit feature, which provides a gentle peak-limiting algorithm so you can record at high levels without worrying about clipping. The 12-segment left and right LED meters have additional indicators to reflect a user-defined "over" (which can be from one to four consecutive full-scale samples).

Frequency response is rated at 10 Hz to 20 kHz (±0.025 dB, at 44.1 kHz) and dynamic range at 120 dB. THD + N is rated at -112 dB. Apogee Electronics Corp.; tel. (310) 915-1000; fax (310) 391-6262; e-mail info@apogeedigital.com; Web www.apogeedigital.com.

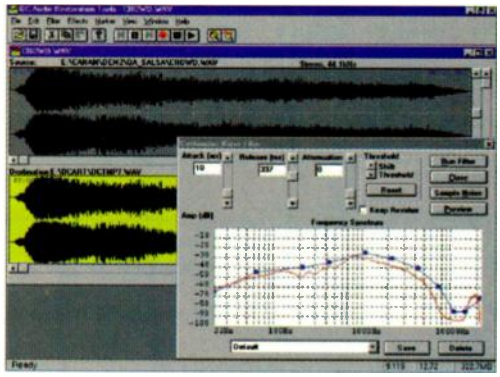
*Circle #412 on Reader Service Card*

## TRACER DIAMOND CUT ART 32

Loaded with features, the 32-bit version of *Diamond Cut Audio Restoration Tools* (Win; $199) from Tracer Technologies can spruce up old recordings and save them as WAV files. This version has many new tools, and all functions can be performed in real time.

*DC-ART 32*'s processing tools give you a range of controls for cleaning up noisy tracks and enhancing dull audio. The long list includes a harmonic exciter, expander/gate, compressor, de-esser, 10-band paragraphic EQ, spectrum analyzer, and time expansion and compression. One of the new features is Virtual Valve Amplification, which simulates seven tube types. Also included is the Mark Silent Passages tool, which identifies spaces between songs and automatically divides your file into several smaller WAV files for CD mastering.

To help you make use of this vast array of tools, *DC-ART 32* comes with thorough documentation, including several Help screens, a 300-page manual with tips and tutorials, and a dedicated Web site (www.enhancedaudio.com) with additional information.



Minimum system requirements are a '486 with 16 MB RAM and Windows 95, 98, or NT, but a Pentium 133 or better is required if you want real-time operation. Tracer Technologies, Inc.; tel. (717) 843-5833; fax (717) 843-2264; e-mail info@tracertek.com; Web www.tracertek.com.

*Circle #413 on Reader Service Card*

## COOL STUFF LABS GENERATOR X

If you're still using an analog signal generator with your digital audio workstation, or even a 16-bit test-tone CD with your 24-bit setup, you may want to take a look at Cool Stuff Labs' *Generator X* ($70), a downloadable plug-in for AudioSuite and TDM.

Providing a 24-bit digital signal generator with 48-bit internal precision, *Generator X* can produce sine, square, triangle, and sawtooth waves, as well as white and pink noise. The plug-in's Single Impulse mode lets you audition reverb sounds without coloration. Rectification, phase inversion, and bypass features are also offered.

*Generator X*'s graphic interface allows you to see and adjust input gain, frequency, and amplitude values, and its


input and output meters let you monitor peak level. A triggering section offers controls for threshold, attack, and release and can key the generator from the input or from a sidechain.

Cool Stuff suggests that, beyond obvious applications such as using the plug-in for alignment or reference tones, you can use *Generator X* for simple additive synthesis or to add tonal emphasis beneath kick drums.

*Generator X* requires at least a 680X0 Mac and a TDM- or AudioSuite-compatible host application. The plug-in is not copy protected and is available only as a download from the company's Web site. Cool Stuff Labs; tel. (650) 366-8648; fax (650) 568-3236; e-mail info@coolstufflabs.com; Web www.coolstufflabs.com.

*Circle #414 on Reader Service Card*

## WAVES L2 ULTRAMAXIMIZER

Waves has introduced the L2 Ultramaximizer hardware master limiter ($1,995). This new 2-rackspace device, which evolved from the company's *L1 Ultramaximizer* software, has 48-bit internal processing and supports 96 kHz sampling rates.

The L2 features a digital limiter with an adjustable threshold and output ceiling from 0 to -30 dBFS. Release time can be adjusted manually from 0.1 to 1000 ms, or you can switch each channel independently to Auto Release Control (ARC), which automatically and continuously selects optimal release times for maximum level output with minimum distortion.

Operable in either stereo or dual mono, the L2 offers balanced analog I/O on XLR connectors, as well as S/PDIF on RCA jacks and AES/EBU on XLRs. The unit supports sample rates of 44.1, 48, 88.2, and 96 kHz and includes A/D and D/A converters with 24-bit internal resolution. Sync options include internal clock, word clock, or synching to either digital input.


The L2 applies Waves' proprietary Increased Digital Resolution dithering technology, boasting transparent word-length reduction with seventh-order noise shaping. Two master dither choices are available, as well as three noise-shaping curves, with output selectable to 24, 22, 20, 18, or 16 bits.

The L2's front panel sports dedicated knobs for threshold, ceiling, and release for each channel, each with a numeric LCD and peak-hold LED ladder. Buttons with associated LEDs indicate selections for signal input, sync input, sample rate, sample-rate doubling (44.1 to 88.2 kHz, or 48 to 96 kHz), output word length (quantizer), dithering type, noise-shaping curve, stereo channel linking, bypass mode, and ARC on/off. Waves; tel. (423) 689-5395; fax (423) 688-4260; e-mail sales-info.us@waves.com; Web www.waves.com.

*Circle #415 on Reader Service Card*

# NOW HEAR THIS

## ALESIS

Alesis's first powered monitor, the M1 Active ($649), features two custom drivers, a 75W woofer amp, 25W tweeter amp, and high- and lowpass active crossover filters with a crossover point of 1.5 kHz and a 48 dB/octave cutoff. Time-alignment circuitry in the tweeter section of the crossover ensures that the high and low frequencies arrive simultaneously.

The M1 Active's 6.5-inch shielded woofer is made from nonwoven carbon fiber with a Santoprene surround, while the 1-inch tweeter dome is made of silk. The tweeter features an internal pole-piece-mounted phasing plug designed to lower the distortion and smooth out frequency response, and it is ferrofluid cooled. Input is on a balanced ¼-inch/XLR combination connector.

Alesis rates the M1 Active's frequency response at 50 Hz to 20 kHz (±2 dB). For the low-frequency amp, total harmonic distortion is rated at <0.03% and S/N ratio at >110 dB (both measurements A weighted). THD for the high-frequency amp is rated at <0.06% and S/N ratio at >112 dB (both at rated output). Alesis Corporation; tel. (800) 5-ALESIS or (310) 255-3400; fax (310) 255-3401; e-mail alecorp@alesis1.usa.com; Web www.alesis.com.

*Circle #416 on Reader Service Card*

## NHTPRO

NHTPro has introduced three monitor systems. The A-10 ($1,200/pr.) and A-20 ($1,900/pr.) systems ship with a pair of monitors and an accompanying power-control unit, while the M-00 ($350/speaker) is an active monitor with built-in amp and heat sink. All three systems are designed to exhibit a detailed resolution, flat frequency response, good off-axis response, and negligible distortion.

The shielded A-10 uses a 6.5-inch, treated paper woofer and a 1-inch soft-fabric dome. It can deliver an SPL of up to 116 dB and has a self-noise of <10 dB (A weighted). The A-10's frequency response is rated at 55 Hz to 20 kHz (±2 dB), and THD is rated at <0.8% (at 90 dB SPL; 100 Hz to 10 kHz at 1m). Its amplifier provides 150W per channel and features a five-position sensitivity switch, power and clipping indicators, and a crossover control.

The A-20, which is also shielded, features an improved woofer and uses a 1-inch aluminum dome tweeter. This system's peak output is 117 dB SPL, and self-noise is <10 dB, but its THD is <0.4%. Frequency response on the A-20 is rated at 48 Hz to 20 kHz (±2 dB). The A-20's amplifier provides up to 250W per channel and, in addition to having the same features as the A-10 amp, boasts a three-character LED that indicates SPL, output device temperature, and line voltage. A-10 and A-20 have balanced ¼-inch TRS and balanced XLR connectors.

The active, shielded M-00 speaker has a 4.5-inch treated-paper woofer and 1-inch soft-fabric dome, with a built-in 75W amplifier. It has a peak output rated at 111 dB, self-noise at 20 dB, and THD at <1.0%. The M-00 has unbalanced RCA and balanced XLR and TRS connectors. NHTPro; tel. (707) 751-0270; fax (707) 751-0271; e-mail customerservice@nhtpro.com; Web www.nhtpro.com.

*Circle #417 on Reader Service Card*

## HAFLER

Hafler has just introduced the TRM6 two-way, biamped, active monitor system ($1,399/pair), which is similar in design to the 1999 **EM** Editors' Choice Award–winning TRM8 but in a smaller, lower-priced package. Its companion, the new TRM10S powered subwoofer ($695), is also compact, measuring 1 cubic foot; it can be used to enhance any monitor system but is designed to complement the TRM6.

The TRM6 system has 1-inch soft-dome tweeters along with magnetically shielded, 6.5-inch polypropylene woofers. A combination ¼-inch/XLR jack and an RCA jack are located on the back of each speaker, as are switches for input sensitivity, tweeter and woofer mute, bass and treble shelving, and balanced or unbalanced input. LEDs on the front of the enclosure indicate power, clipping, and overheating.

As in the TRM8, Hafler builds its Trans•nova amplifier technology into the TRM6, giving it a 33W amp for the tweeter and a 50W low-end amp. The adjustable crossover is a 24 dB/octave Linkwitz/Riley with 40 Hz to 110 Hz range. The system's peak output is at least 120 dB SPL. Frequency response is rated at 55 Hz to 21 kHz (±2 dB) and THD at 0.5% (100 Hz to 21 kHz).

The TRM10S subwoofer has a 10-inch speaker in a vented enclosure and can extend down to 25 Hz. The unit has both XLR and RCA inputs and a 200W class G Trans•ana amp. Input sensitivity is adjustable with a gain/potentiometer pot. Another continuously variable pot sets the 24 dB/octave crossover point between 40 Hz and 100 Hz. Hafler; tel. (888) HAFLER1 or (602) 967-3565; fax (602) 894-1528; e-mail sales@hafler.com; Web www.hafler.com.

*Circle #418 on Reader Service Card*

# PERFORMANCE TOOLS ▲ ▲ ▲ ▲

## ▲ ALLEN & HEATH

Allen & Heath has released the 10-input, 4-output DL1000 digital mixer ($1,395) and its 300W/channel powered counterpart, the DP1000 ($1,595). The two mixers are the first in the company's new Icon series, designed for use in live performance.

The mixers have six mono mic/line channels, each with balanced XLR and unbalanced ¼-inch inputs. In addition, they have two mono mic/stereo line channels on unbalanced ¼-inch connectors. The XLR inputs provide 48V phantom power.

Channels 1 through 6 have trim controls, whereas Channels 7 and 8 have a trim knob for each stereo pair. A switch enables the user to toggle between the mono mic preamps and the stereo line-level inputs for Channels 7 and 8. Each channel has dedicated controls for the EQ, solo, and mute; a noise gate; a compressor; and 4-band EQ with parametric mid bands and sweepable high and low bands.

Both mixers have 10-band graphic EQ and two stereo effects processors with more than 80 adjustable presets. The ADCs and DACs are 20-bit.

Users can store their settings in Song patches, which can be sequenced for a performance and triggered by using an optional footswitch, a dedicated front-panel button, or a MIDI controller. Parameters can be viewed on the mixer's large LCD.

Allen & Heath rates the mixers' frequency response at 20 Hz to 20 kHz (+0/-1 dB), dynamic range at 91 dB (A weighted), channel crosstalk at <-90 dB, and THD at <0.008%. Allen & Heath; tel. (801) 568-7660; fax (801) 568-7662; e-mail customer@dbxpro.com; Web www.allen-heath.com.

*Circle #419 on Reader Service Card*

## ▼ FENDER

Fender's new line of amplifiers features the company's ground-breaking Stereo Field Expansion (SFX) technology. The SFX amp splits the stereo signal into sum and difference signals, which are played through very closely placed speakers that are out of phase by 180 degrees. The sound waves combine acoustically around the cabinet to form a perceived stereo field of up to 300 degrees. Each of the three new amp models also comes with 32 onboard stereo digital effects, including reverb, chorus, delay, and flange.

The Acoustasonic SFX amp ($899.99) is designed for live instrument and vocal performance. Of the amp's two channels, the first is optimized for acoustic instruments, attenuating harsh high frequencies that are typical of piezo pickups; the second channel is set up with balanced XLR and ¼-inch inputs for mic- or line-level signals and has separate tone controls and notch filters. This amp also has effects inserts for outboard gear.

The SFX Keyboard 200 amp ($999.99) has two stereo channels, each with left and right inputs, and a third, mono channel for mic or line-level input via balanced XLR and ¼-inch jacks. Each channel has separate 3-band EQ controls. The amp has a subwoofer line out and ¼-inch line inserts.

Both the Acoustasonic SFX and the SFX Keyboard 200 amplifiers deliver 80 watts per channel, and they include Fender's innovative three-way SFX speaker array.

To give any amplifier the SFX spatial power, the SFX Satellite ($749.99) can be added to an amp's effects loop. The Satellite has the 32 DSP effects presets and adds 80 watts to your system, as well as a specially designed 12-inch speaker that enables your rig to produce the SFX stereo spread. Fender Musical Instruments; tel. (602) 596-9690; fax (602) 596-1384; Web www.fender.com.

*Circle #420 on Reader Service Card*

## ELECTRO-VOICE

New from Electro-Voice is the Sx80 speaker system ($530/pair), the latest addition to the company's System 2000 series. This system houses the same technology found in the E-V Sx300 and Sx500 speakers but in much lighter, ultracompact cabinets.

The Sx80 enclosure is only 15.75 (H) x 11.5 (W) x 8.75 (D) inches and weighs 16 pounds, making it very useful to traveling musicians. Each contains an 8-inch woofer and a 90 x 65-degree high-frequency horn. The system uses E-V's Ring-Mode Decoupling technology to eliminate unwanted resonance from the vibrations of the speaker enclosure and internal components. The amp delivers 175 watts per enclosure. Each Sx80 provides a peak output of 120 to 125 dB SPL. Frequency response is rated at 70 Hz to 20 kHz (-3 dB).

These speakers come with a white or paintable black cabinet finish. In order to accommodate the needs of all users, the Sx80 system will soon be available in a variety of styles, including a transformer version, a weather-resistant model, and with various mount types; check with Electro-Voice for availability and prices. Electro-Voice; tel. (800) 234-6831 or (616) 695-6831; fax (616) 695-1304; Web www.electrovoice.com.

*Circle #421 on Reader Service Card*



32 x 8 Channel Digital Mixer with the Human Touch

The Spirit Digital 328 represents a refreshing departure in digital console design, retaining the ease of use of a conventional analog console, yet providing all the advantages of 24-bit digital. The 328 is nothing like a computer with faders. The key to the 328 is the unique "E-Strip", which avoids the need for tedious menus and brings instant access to all 16 channel inputs, 16 tape returns, auxiliary sends and returns, EQ and effects for each channel. Included as standard are two on-board Lexicon effects processors, two dynamic processors, Tascam TDIF® and Alesis ADAT® optical interfaces and a built-in meterbridge, with no hidden "options" to add to the cost. If you want the functionality of a digital console, but the common sense approach of an analog 8-bus board, you need to check out the Spirit Digital 328. It's a refreshing change!

## Main Features

- 16 Mic/Line Inputs
- 8 Group Outputs
- Up to 42 Inputs at Mixdown
- 2 ADAT Optical Interfaces
- 2 Tascam TDIF Interfaces
- 3-band Parametric EQ
- AES/EBU Interface
- UltraMic+ Preamps w/ 66dB of gain
- Instant Recall Capability Recording
- Analog Console "Feel"
- 2 Lexicon Effects Processors
- 16 Digital Tape Returns
- 5 Pairs of Stereo Inputs
- 24-bit A/D and D/A Converters
- 128x Oversampling Throughout
- Snapshot Automation with 100 Scenes
- Built-in Meterbridge
- SP/DIF Interfaces
- 100mm Motorized Faders
- 2 Units may be Cascaded for 32 Tracks

**$4,999.95 U.S. Retail**

SPIRIT
by Soundcraft

Spirit By Soundcraft, Inc.
4130 Citrus Ave., Suite 9
Rocklin, CA 95677
Tel: (916) 630-3960 • Fax: (916) 630-3950

A Harman International Company

www.spiritbysoundcraft.com

circle #523 on reader service card

## ▶ A.R.T. DMV-PRO

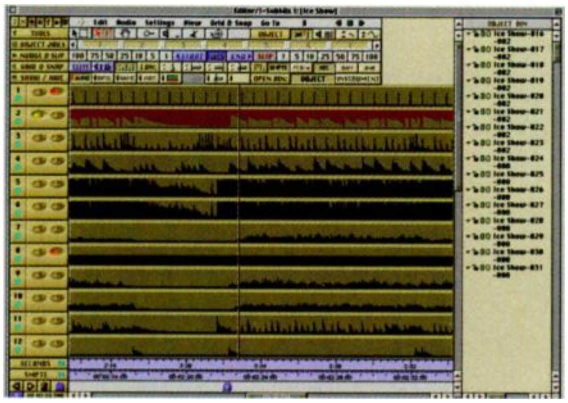

The DMV-Pro from A.R.T. ($499) is a dual discrete-stereo digital multi-effects processor capable of processing two independent stereo signals simultaneously, each with up to three effects at a time. Each stereo channel has an edit knob for adjusting parameter settings and a 2-line, 16-character LCD to display effects-edit status. A global knob scrolls through presets, and a 3-character LED shows the preset number. All DMV-Pro parameters can be adjusted via MIDI in real time.

The DMV-Pro offers 72 effects algorithms, such as reverbs; chorus; flange; pitch shift; tremolo; rotary-speaker simulator; delay; and more. Up to 12 parameters are available for each effect. The unit has 100 user preset locations, which come loaded with factory presets.

The DMV-Pro's dynamic controllers adjust effects settings depending on input signal strength (that is, the effect depth increases as the signal level increases). The unit's smart-encoder features allow you to tap in delay time, repeat hold, and rotary-speaker simulation speed.

The A/D converters are 20-bit, 64x oversampled delta-sigma; the DACs are 20-bit, 128x oversampled. The DMV-Pro uses 24-bit internal processing. Input and output are on ¼-inch unbalanced connectors. A.R.T. rates the unit's dynamic range at >93 dB and THD at <0.01 @ 1 kHz. An included 9 VAC adapter powers the unit. A.R.T.; tel. (716) 436-2720; fax (716) 436-3942; e-mail artroch@aol.com; Web www.artroch.com.

*Circle #422 on Reader Service Card*



## ▲ ENSONIQ PARIS CONCEPT

Ensoniq has released two new entry-level versions of the company's PARIS. The PARIS Concept ($1,299.99) and Concept FX ($1,799.99), both of which are cross-platform (Mac/Win), ship with the same software as PARIS, and allow you to record and edit up to 16 tracks of 16- or 24-bit digital audio. You can play back up to 128 tracks on either version.

PARIS Concept consists of an EDS-500 PCI card and Interface 2, a 2-channel audio interface that offers 20-bit, 128x oversampling ADCs and 20-bit DACs on TRS ¼-inch connectors. The Concept FX package uses the Interface 2 and an EDS-1000, which delivers real-time 24-bit effects.

The PARIS software offers four bands of full-bandwidth parametric EQ per channel, pan control, solo, channel muting, phase inversion, and real-time automation. DSP functions include gain change, normalize, polarity inversion, sample-rate conversion, time compression and expansion, and pitch shifting.

You can also run third-party DirectX and VST plug-ins on the Windows version and VST plug-ins on the Mac. The PARIS Concept FX system adds a number of real-time, 24-bit effects, including compression, expansion, gating, chorus, delay, and reverb.

In addition to the PARIS software, the systems are bundled with Steinberg's *WaveLab Lite* for Windows, BIAS's *Peak LE* for Mac, and Prosoniq's *SonicWorx Essential* for Mac (a special PARIS version of *SonicWorx*). Each system requires at least a Pentium 133 or Power Mac 604/120 with 48 MB RAM (64 MB recommended). Ensoniq Corp.; tel. (800) 553-5151 or (610) 647-3930; fax (610) 647-8908; faxback (800) 257-1439; Web www.ensoniq.com.

*Circle #423 on Reader Service Card*

## ▼ LYNX STUDIO LYNXONE

Lynx Studio Technology's LynxONE ($549) is a PCI audio card that offers up to four simultaneous channels of 24-bit recording and playback. The card can operate at +4 dBu or -10 dBV.

The LynxONE provides two channels of analog I/O and one stereo pair of AES/EBU or S/PDIF digital I/O (switchable), all on XLR connectors. The analog ports support 8-, 16-, 20-, or 24-bit recording and playback with adjustable sampling rates from 8 to 50 kHz. The digital ports support 16-, 20-, or 24-bit recording and playback at 32 kHz, 44.1 kHz, 48 kHz, 88.2 kHz, or 96 kHz. The digital and analog I/O are simultaneously available for 4-channel operation. A two-foot breakout cable (included) provides two pairs of MIDI In/Out ports for 32-channel operation.

The LynxONE features a field-programmable gate array, which is optimized for record monitor mixing or for using the card as a standalone A/D or D/A converter. You can sync the card via word clock or video clock, or with other LynxONE cards.

Dynamic range is rated at >103 dB, and THD+N is <0.0025%. The LynxONE requires a Pentium 90 or DEC Alpha processor with a free PCI slot and minimum 16 MB RAM. Lynx ships the card with Windows 95, 98, and NT drivers. Lynx Studio Technology; tel. (949) 515-8265; fax (949) 645-8470; Web www.lynxstudio.com. ◉

*Circle #424 on Reader Service Card*

# Start To The Closer Top

**Digidesign**

# ToolBox

**ToolBox for Windows***

Audiomedia III PCI card
Session Software
Logic Audio AV from Emagic
Sound Forge XP from Sonic Foundry
Acid Rock from Sonic Foundry

**A $1349 value for only $549!**

**ToolBox for Mac OS**

Audiomedia III PCI card
Pro Tools PowerMix Software
D-fx & D-Fi Plug-In Bundles
SFX Machine Lite from Bias
Peak LE from Bias

**A $2400 value for only $895!**

More audio professionals use a Digidesign digital audio workstation (DAW) than any other. That means, no matter what you buy today, you'll probably end up on a Digidesign system tomorrow.

Why not give yourself a head-start with the Digidesign ToolBox for Mac OS or Windows 95/98/NT?

This incredibly affordable bundle includes everything you need to build your own, professional-sounding audio productions. And it's upgradeable to Pro Tools|24, so your investment is protected well into the future.

Need a DAW? Start out with the leader, and stay ahead of the game.

(Pictured: Audiomedia III PCI Card)

**Upgradeable** to our award-winning Pro Tools|24 total audio production solution.

PRO TOOLS

1.800.333.2137, code 427
www.digidesign.com/toolbox

* Available on Windows 95/98/NT



digidesign
A division of Avid

SEE US AT NAMM BOOTH #W1205

circle #511 on reader service card

PRO FILE


# The Meaning of Live

## Mice Parade strikes up the one-man band.

*By Rick Weldon*

Port Chester, New York's Mice Parade is a project of drummer/composer Adam Pierce. On his latest CD, *The True Meaning of Boodlebaye,* Pierce plays drums, guitars, saxophone, synths, vibes, and even sings an occasional tune. In addition to presenting traditional structures of Western music, the music subtly introduces such styles as Balinese and Javanese gamelan as well as musical elements that, at first listen, sound as though they were sequenced. The album appears on Bubble Core, a label that Pierce and partner Dylan Cristy have operated since 1995.

"All but two of the songs were recorded at my house on an Alesis ADAT," remarks Pierce. "I also have a moody TEAC 3340 reel-to-reel and a Mackie MS1402-VLZ mixer, and the entire record was recorded with two Shure SM58 mics. I live with four other musicians: all told, we have 3 Fender Rhodes electric pianos, 25 analog and digital synths, 2 sets of vibraphones, 3 drum kits, violins, harps, ukuleles, *charangos,* and a variety of guitars and percussion. It's really an amazing situation that none of us takes for granted.

"I've also collected a few effects, like my Roland Chorus Echo, and some handmade stuff like the 'Radam unit,' made by my housemate Shane DeBlasio."

"The Radam unit is, of course, named after Adam, who's rad," explains DeBlasio. "While it's better known as a ring modulator, in this case that's a misnomer. They were called 'ring modulators' in the '60s because that's what they were: there was a ring of diodes. But the Radam is based on Craig Anderton's design, from his book *Electronic Projects for Musicians.*"

According to Pierce, "The first thing you hear on the record is a single note. It goes on and develops into more and more notes, but I played only one note at a time on the keyboard. That lovely ring modulator is filling in the chords."

Although *Boodlebaye* contains no sequencing, the song "Headphonland: The Gangster Chapter" brought Pierce a step closer to breakbeat convention. "The one-measure hip-hop beat that's in there is a sample," he says. "I used a Roland MS-1 sampler, retriggering the pattern in time with the rhythm."

Flurries of electronica-style percussion also pop fleetingly into the mix, especially on the lightning-fast breaks of "Surprise! The Slippery Sled Went Down" and "Shalom." "Those were done with a *caja,* a Spanish wooden box traditionally used in flamenco music. It doesn't have a thudding drum sound, just a snap, and I would layer that on top of the live kit sound.

"I'm interested in mimicking drum machines and sequencers, while at the same time recording everything in one take," says Pierce. "Every track, and every mix on the album, is done in a single take. I don't believe in a preconceived notion of perfection, where, if you don't get exactly what you were planning for, you didn't get it 'right.' It's also a challenge to do well with one take. It forces you to get creative."

Because he's a drummer, rhythmic creativity is especially important to Pierce. "In my opinion, much of today's popular music is contributing to our society's decline, reinforcing TV, McDonald's—everything that is easy and nice and slothful. When I listen to music from other parts of the world, I hear so much more going on. I think it's fascinating when people are brought up to feel any rhythm naturally."

*For more information, contact Bubble Core; tel. (914) 939-7717; e-mail thehouse@bubblecore.com; Web www.bubblecore.com.*

**Adam Pierce is Mice Parade.**

# TECH PAGE

# Chaotic Computing

## *Chaos theory: a new computing paradigm?*

*By Scott Wilkinson*

I've said it before, and I'll say it again: any advancement in computer technology will ultimately have a direct impact on electronic music. After all, synthesizers, signal processors, and most of the other tools of our trade are essentially computers dedicated to the task of creating or modifying musical sounds. If computers can be made faster, smaller, and/or cheaper, then electronic musicians will benefit.

Among the new and esoteric approaches to computing is a paradigm proposed by William L. Ditto of the Georgia Institute of Technology and Sudeshna Sinha of the Institute of Mathematical Sciences in Madras, India. This paradigm is based on *chaos theory*, a relatively new branch of physics that analyzes the behavior of certain types of systems.

A simple example of a chaotic system is a magnetic pendulum that is suspended over another magnet and allowed to swing in any direction. The two magnets are oriented so that like poles face each other. If you pull the pendulum away from the center and let go, it moves toward the stationary magnet, which repels the magnet in the pendulum, sending it in a new direction. Each time the pendulum swings back toward the center, it careens in a new direction.

At first glance, this motion might seem completely random, but it isn't. It is called *chaotic motion*, and it can be described mathematically. In fact, if you plot the position of the pendulum over time, a pattern emerges. This pattern is called a *strange attractor*, of which there are several types (see **Fig. 1**).

Ditto and Sinha have demonstrated that an array of coupled chaotic elements can perform various basic mathematical operations, such as addition, multiplication, and Boolean algebra. The elements can consist of almost any chaotic system, such as certain types of lasers or silicon/neural-tissue hybrid circuitry.

The computational elements are connected into a lattice using an adaptive mechanism that opens when a particular element reaches a critical value. This resembles the way in which neurons in the brain work together. The mechanism is designed to allow a wide range of critical values to vary the connections between elements.

Initial values can be encoded in several ways: as chaotic patterns, waveform amplitudes, or the frequency of spikes in the chaotic behavior. After these values are encoded, the elements are stimulated to interact, which causes intermediate values to "cascade" within the lattice. These cascades eventually evolve into the answer to the initial problem.

According to Ditto, "We are not setting up rules in the same sense that digital computers are programmed. The system develops its own rules that we are simply manipulating. We don't micromanage the computing; we let the dynamics do the hard work of finding a pattern that performs the desired operation."

Like any technology, a chaotic computer will have its own strengths and weaknesses. "It might be better for those activities that digital computing doesn't do very well, such as pattern recognition or detecting the difference between two pieces of music," Ditto says.

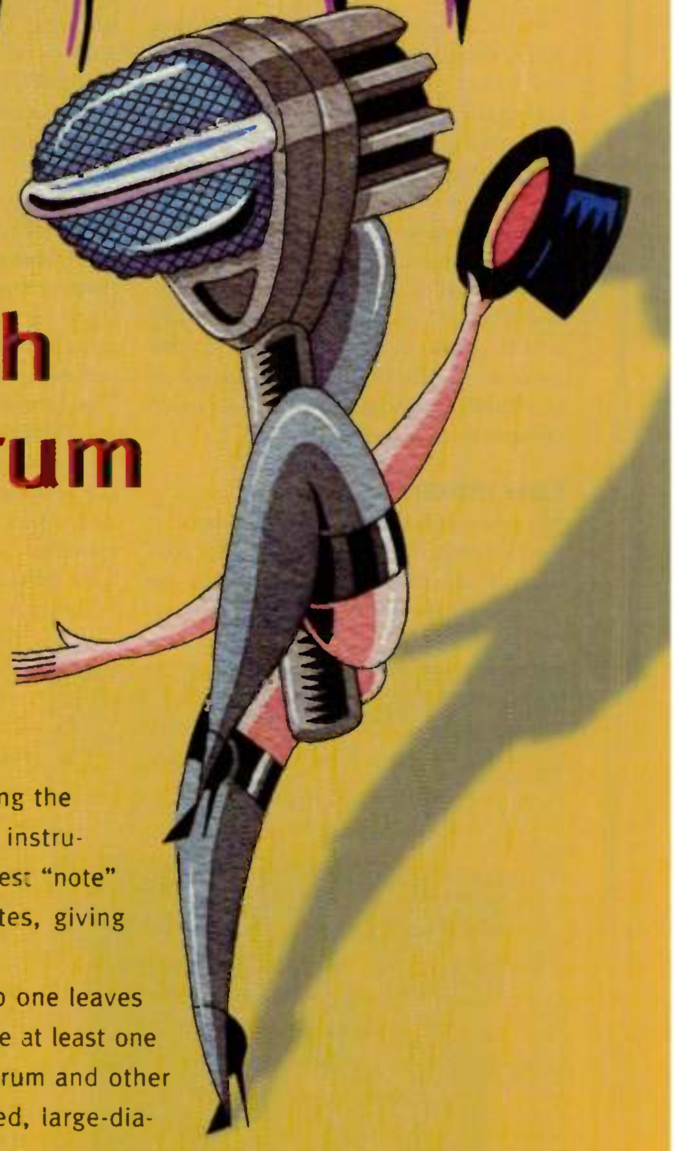
He believes the new approach could work especially well with optical elements: "Potentially, we could stimulate a very fast system of coupled lasers to perform highly complicated operations, like Fourier transforms." Ditto envisions a system using lasers that generate femtosecond (millionth of a nanosecond) pulses to perform billions or even trillions of calculations per second.

Of course, chaotic computing is in its infancy, and many substantial hurdles must be overcome before it can be applied to practical applications, such as electronic-music systems. But the potential is exciting, and I look forward to following the development of this technology. ❂

**FIG. 1: Chaotic systems conform to special patterns called strange attractors, such as this one, the Lorenz attractor.**

Illustration by Paul Moch

# Kickin' It



## Get your studio kicks with seven bass-drum microphones.

A killer bass-drum sound is an integral part of a killer mix. It's the foundation, the meat of the meat and potatoes (snare drum being the potatoes), the beast of burden that the other instruments ride on. The kick drum, typically the lowest "note" in a mix, sits just beneath or "inside" the bass notes, giving substance, impact, and definition to the low end.

Understandably, capturing that low note is not a job one leaves up to just any microphone. Manufacturers often dedicate at least one microphone in their product lines to recording bass drum and other low-frequency sound sources. This is usually a rugged, large-diaphragm dynamic mic with high-SPL handling and a frequency response tailored to complement or enhance the sound of the drum.

Over the years, a handful of dynamic microphones have distinguished themselves as exceptional bass-drum mics, including the AKG D 112 and its predecessor, the D 12; the Electro-Voice RE20; the Sennheiser MD 421 (also known as a great tom mic); and the Beyerdynamic M 88. (Some engineers prefer using condenser miccrophones for recording kick drum, but that's a different story.) In recent years, the variety of bass-drum mics on the market has expanded considerably. Some of the above-mentioned companies have come out with new models, and several other manufacturers have added bass-drum mics to their catalogs.

By Brian Knave with Myles Boisen


With all those kick mics to choose from, what's the prospective buyer to do? Short of purchasing one of each and taking them into the studio for a spin, there's little opportunity to compare them. Even the best-equipped music stores don't provide the proper environment and necessary gear for assessing how well a microphone records, especially when it comes to drum mics. For that reason—not to mention the fact that **EM** currently has four drummers on its editorial staff—we thought it was time that someone did a critical comparison of kick-drum mics.

## LOW RIDERS

We researched the market and turned up 11 dynamic mics designated for bass drum. Prices ranged from $120 on up, so we leveled the field somewhat by considering only those priced between $250 and $400, thereby eliminating the two lowest-priced models. Subsequently, two manufacturers chose not to participate in the comparison tests. This left us with seven microphones: the AKG D 112, Audio-Technica ATM25, Audix D4, Beyerdynamic TG-X 50, Electro-Voice N/D868, Sennheiser E602, and Shure Beta 52.

Even then, the playing field still was not exactly level because the manufacturers employ different designs for their bass-source microphones and aim at somewhat different markets. The Electro-Voice N/D868, for example, is labeled "Bass Drum" right on the mic, and the accompanying literature describes it as "designed specifically for bass drums." The AKG, Beyerdynamic, Sennheiser, and Shure mics, however, are described more generally as being suitable for low-frequency sources, including kick drum, bass-guitar cabinets, and so on.

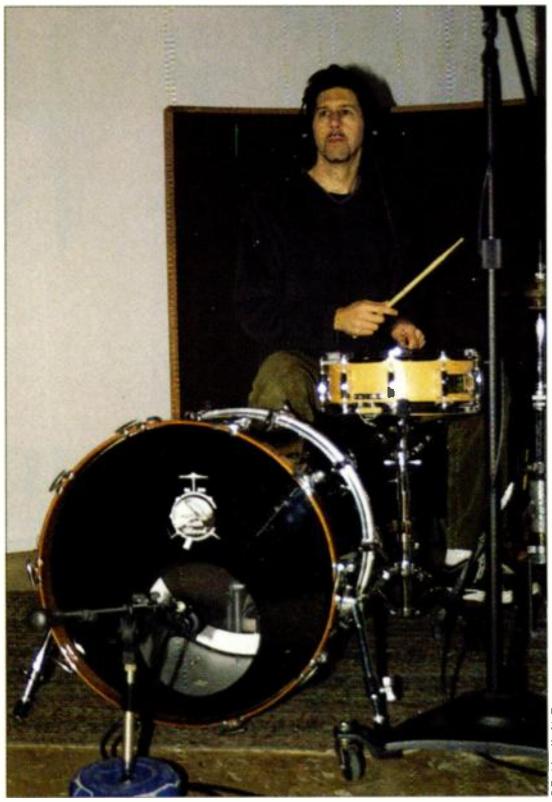

BRIAN KNAVE

**FIG. 2: Hanes's 22-inch Gretsch drum sounded huge in the live room at Guerrilla Recording.**

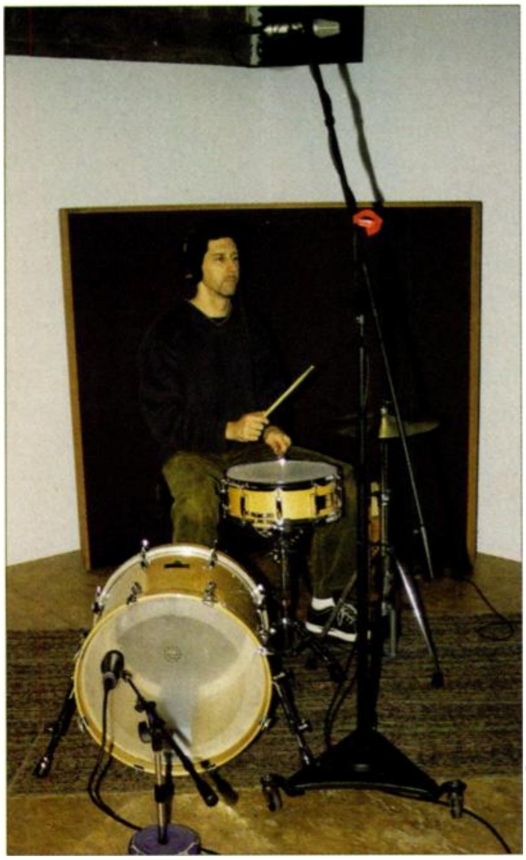

BRIAN KNAVE

**FIG. 1: Drummer John Hanes kicks off EM's bass-drum mic tests playing his 20-inch Pearl Masters Custom bass drum.**

An application booklet that comes with the Audix D4 specifies kick drum, floor tom, and bass-guitar cabinet, but extends the applications to djembe and Leslie bass cabinet. Audio-Technica describes the ATM25 as well suited for "drums (kick, tom, snare), timpani, piano, acoustic and electric bass, trombone [and] vocal pickup where low-frequency emphasis is desirable."

These distinctions may seem negligible, but they proved relevant in our comparisons. Although we tested the seven mics on bass drums only, it became evident that, in terms of alternate applications, some of the mics were more versatile than others—or perhaps even better suited to other instruments than to bass drums.

It should be noted, too, that each of the mics is intended for live sound as well as studio applications. However, our tests were performed in the studio only (two studios, actually).

## SISTER ACT

Our first step was to request two identical mics from each manufacturer; that way, we could compare each pair for consistency. Also, if one of the mics proved defective—a situation that has arisen in previous microphone comparisons—we would be able to proceed with the tests using the good mic.

To check the mic pairs for consistency, we recorded mono program material to two separate tracks—one through each microphone—and then compared the results. I also talked and sang through the mics. Within each pair, the mics sounded virtually identical, except for the two ATM25s, which sounded quite different from each other. (As it turned out, one of them was evidently damaged during shipping.)

## METHOD TO THE MADNESS

We performed the comparison tests both at my own personal studio and at Guerrilla Recording in Oakland, California, a professional facility co-owned

For manufacturer information, please see Contact Sheet, p. 227.

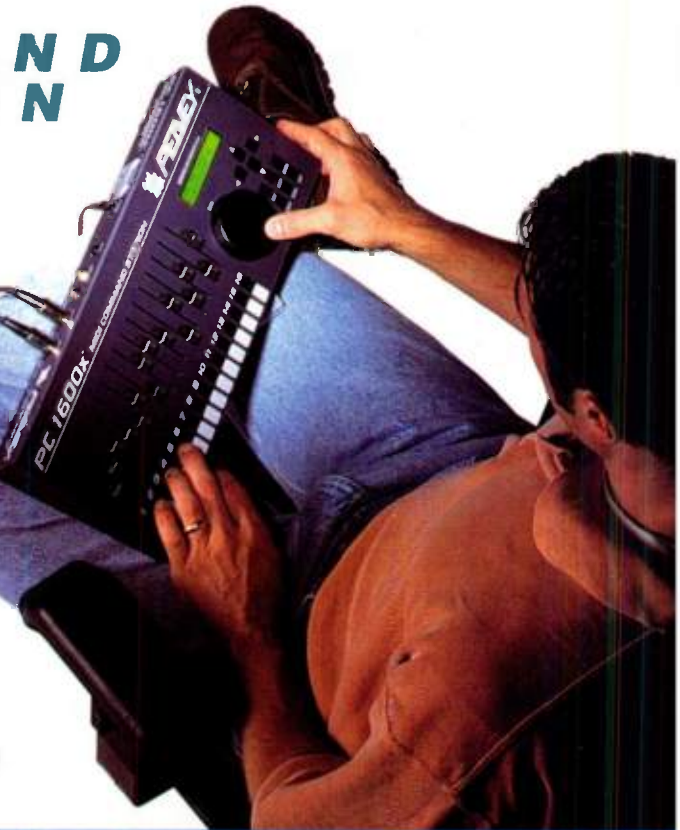

# Ultimate Control For Control Freaks!

The PC 1600x™ is the ultimate MIDI controller. If you're sick of editing your events one at a time, put away your mouse, sit back in your chair, and control everything with the PC 1600x.

Originally designed to work as a keyboard controller, our customers quickly discovered that the PC 1600x can control any MIDI device. With 16 faders, 16 buttons, and 2 CV/footswitch inputs, ***its applications are limitless.***

# PC 1600x™

## MIDI COMMAND STATION




***Theme Parks*** - Control special effects, lighting, sound systems.

***PCs*** - Control hard disk recorders, sequencers, or sound cards.

***Lighting*** - Control anything that accepts MMC.

***Recording Studios*** - Control all tape recorders, minidiscs, and DAW systems that accept MIDI commands (example: Digidesign® Pro-Tools™, and Cakewalk™).

***Synthesizers*** - Real-time control of virtually any MIDI parameter in today's synths. Edit patches with no extra hardware. No PC needed. Real-time control of FX processors.

***Video Production*** - Control/edit VTRs.

***Computer Animation*** - Intuitive keyframe control for character animation

SIDE EFFECTS SOFTWARE Houdini · cakewalk · digidesign DEVELOPMENT PARTNERS

**Virtually anything that's MIDI can be controlled by the Peavey PC 1600x MIDI Command Station!**

Extended Warranty · Peavey Electronics

VISIT US ON THE WEB www.peavey.com

Peavey Electronics Corporation • 711 A Street • Meridian, MS 39301 • U.S.A. • (601) 483-5365 • Fax (601) 486-1278
Peavey Europe • Great Folds Road • Oakley Hay • CORBY • Northants NN18 9ET • England
Tel: +44 (0) 1536 461234 • Fax: +44 (0) 1536 747222 • ©1998

PEAVEY

circle #514 on reader service card

SEE US AT NAMM BOOTH #4245

# Now Accepting Applications

# Make Your Guitar Scream

## Use the DDP to get the tone you've always wanted

### Try this for a great guitar sound:

**Gate:**
- Threshold at -42dB
- Ratio at 1:4
- Attack at .1 mSec
- Hold at 40 mSec
- Release at 54dB/Sec
- TCM Time at 3 mSec

**Compressor:**
- OverEasy knee #3
- Auto attack and release - On
- Threshold at -10dB
- Ratio at 4:1
- Gain at 6.5

**Parametric EQ:**
- Band 1: 200Hz Boost 3dB
- Band 2: 8kHz Cut 2dB
- Band 3: 4kHz Boost 5dB
- Tape Saturation Emulation: Warm

**Limiter:**
- Threshold at 0dB
- Attack at .1 mSec
- Release at 130dB/mSec

### Use the DDP to:
- Bypass the inferior A/D converters found in computer based recording cards
- Improve the quality of your tracks by recording "hotter"
- Gate noisy guitar rigs


# Smooth Out Your Vocals

## The DDP lets you set em on top of the mix, or blend em in

### On your next vocal session try this DDP setup:

**Parametric EQ:**
- Band 1: 100Hz Cut 3dB
- Band 2: 25Hz Cut 12dB
- Band 3: 20kHz Cut 11dB
- Tape Saturation Emulation: Light

**Compressor:**
- OverEasy knee #4
- Auto Attack/Release: On
- Threshold at -17dB
- Ratio at 2:1
- Gain at 3.5

**Gate:**
- Threshold at -50dB
- Ratio at 1:2.6
- Attack at .1dB/mSec
- Hold at 38 mSec
- Release at 88dB/Sec
- TCM time at 1mSec

**De-Esser:**
- Frequency at 4.6kHz
- Amount at 45%



### The DDP improves vocals by:
- Taming those harsh "esses"
- Controlling vocal levels
- Gating out background noise
- Providing comprehensive vocal processing in a single unit

Save The Planet It's The Only Market We've Got

# Apply Within.

# DDP

# Fatten Up Your Drums

## Get that "fat track" sound you're looking for

### Try this for a great kick sound:

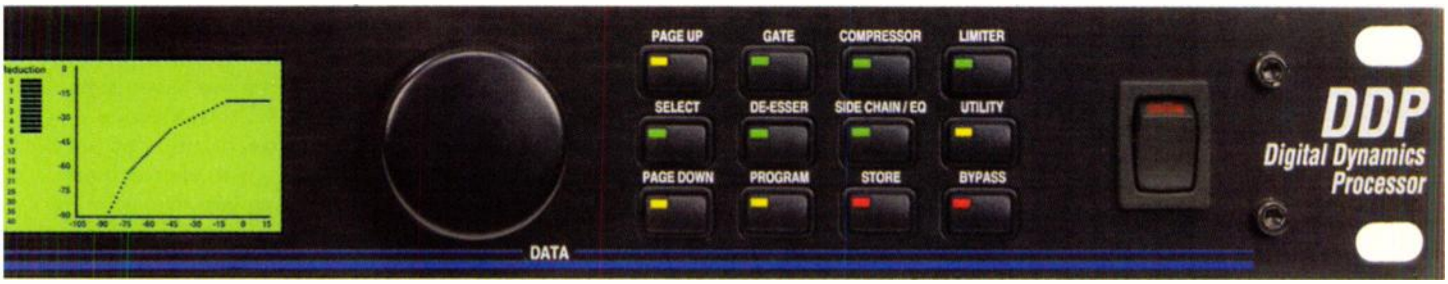
**Gate:**
- Threshold at -32dB
- Ratio at 1:15
- Attack at .4 mSec
- Hold at 12 mSec
- Release at 92dB/Sec

**Compressor:**
- OverEasy knee #3
  - Auto attack and release - On
  - Threshold at -9dB
  - Ratio at 5:1
- Gain at 6.5

**Limiter:**
- Threshold at 0dB
- Attack at .7 mSec
- Release at 120dB/mSec

### DDP is better for drums because:

- Type IV™ allows you to capture the incredible dynamic range of a drum kit
- TCM™ Transient Capture Mode allows more accurate dynamics processing on percussive sounds
- Using the ultra-fast Gate improves drum track separation




# Mix It Down

## Bypass the on-board convertors on your DAT or DAW, and add sensible processing at the same time

### When you're ready to finish it off:

**Gate:**
- Threshold at -60dB
- Ratio at 1:3.5
- Attack at .1 mSec
- Hold at 18 mSec
- Release at 100dB/Sec

**Compressor:**
- OverEasy knee #1
- Auto attack and release - On
- Threshold at -10dB
- Ratio at 2.4:1
- Gain at 7.0

**Limiter:**
- Threshold at 0dB
- Attack at .1 mSec
- Release at 160dB/mSec

### During Mix-down the DDP can:

- Help you mix hotter with no "overs"
- Dither to your final format-16, 20 or 24-bit
- Preserve your stereo image using True RMS Power Summing™
- Add transparent processing to your entire mix

Let's face it. 'Til now there were some audio processing chores (like compression) that just had to be done analog to sound right...right?

Please welcome to your rack the dbx DDP. It's the world's first 2 channel compressor/limiter that has all the warmth, life and mercy of an analog box with the precision converters you, (the modern recordist), MUST HAVE for your digital recording applications.

Finally, a 24 bit digital box that glows with the classic characteristics of dbx® compression that has been processing the hits for over 25 years. Take a run through the parameters, it's REAL easy... you'll see all the standard controls you'd expect, plus quite a few more. 10 steps of OverEasy® on the compressor, not just on/off. Transient Capture Mode™ (TCM), a fully parametric EQ, De-sser and Tape Saturation Emulation™ (TSE) are all standard on the DDP.

All this is hitched to world class 24 bit converters; converters that are found in boxes costing thousands more. Speaking of which, you can pay thousands more, but unless you get a dbx DDP you won't get the dynamic range afforded by our patented TYPE IV™ Conversion System.

Try the DDP on your toughest dynamics applications and see for yourself. Analog sound, Digital tracks, we're taking applications!

dbx PROFESSIONAL PRODUCTS

H A Harman International Company

dbx Professional Products • 8760 South Sandy Parkway
Sandy UT 84070 • Phone (801) 568-7660 • Fax (801) 568-7662
email: customer@dbxpro.com • URL: http://www.dbxpro.com

circle #503 on reader service card

SEE US AT NAMM BOOTH #733


and operated by musician/engineer Myles Boisen, who kindly acted as my assistant. We recorded all tracks one at a time to a prerecorded click track, side by side on the tape, to allow easy comparison through mutes. The tracks at my studio were recorded directly to tape on an ADAT XT through a dbx 1086 mic preamp (with the dynamics section bypassed, of course). At Guerrilla we recorded to 1-inch analog tape (Quantegy 456) on a Tascam MS-16 through a Focusrite Red Series mic preamp. Care was taken to equalize levels during both recording and playback.

Altogether, we recorded five bass drums. At Guerrilla we were assisted by San Francisco Bay Area drummer John Hanes, who played two different drums: a 20-inch Pearl Masters Custom with both heads intact except for a circular pattern of holes punched in the front head (for sonic purposes) and, inside, a rectangle of foam wedged between the two heads; and a 22-inch Gretsch with a large hole cut beneath the center of the front head. For the 20-inch drum, we positioned each mic about two and a half inches from the front head with the capsule aimed directly at the head near the hoop of the drum (see **Fig. 1**). For the 22-incher, the microphones were positioned barely inside the hole and aimed just beneath the beater-impact point (see **Fig. 2**).

In my studio, I recorded three bass drums: a vintage 22-inch Ludwig with the front head removed and a cotton packing blanket lying inside and touching the batter head; a 20-inch Gretsch with a medium-size, off-center hole in the front head and a flannel baby blanket positioned inside the drum and lightly contacting each head; and an 18-inch Slingerland with both heads intact and no muffling except for a strip of felt beneath each head.

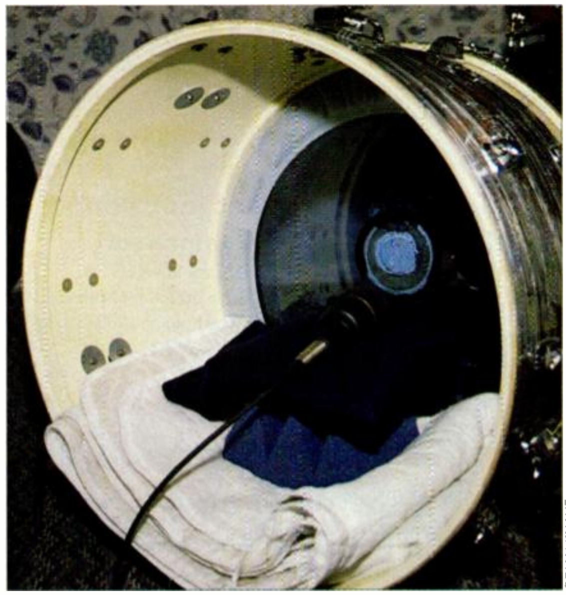

BRIAN KNAVE

**FIG. 4: Positioning the mic inside the kick drum delivers a tighter, more controlled sound with better definition and less resonance. It also reduces leakage from other instruments.**

To accommodate the varying manufacturer-suggested optimal mic positions, I recorded the 22-inch Ludwig drum twice—once with each mic positioned just barely inside the drum and pointed at the beater (see **Fig. 3**), and again with each mic inside the drum, about three to four inches from the batter head and aimed almost directly at the beater-impact point (see **Fig. 4**).

For the 20-inch Gretsch drum, I positioned the microphones just inside the hole, again aimed at the beater (see **Fig. 5**). For the 18-inch Slingerland, I positioned the mics slightly off center and about three inches back from the front head (see **Fig. 6**). I also employed different beaters on the kick pedal: the Slingerland and Gretsch drums were played with a wood beater, and the Ludwig with a standard felt beater.


BRIAN KNAVE

**FIG. 3: Many engineers prefer recording kick drum with the front head of the drum removed—and just as many drummers hate removing it.**

## DRUM-A-RAMA

We used different drums, beaters, mic positions, and rooms, of course, to broaden the scope of the tests. The three drums recorded in my studio, for example, represent the three most commonly used sizes and respective tunings of bass drums. Each is specific to a particular style of music, a different school of thought on tuning, and a different approach to recording. Together with the two bass drums recorded at Guerrilla, this selection is fairly representative of the various sounds an engineer is likely to come across when recording bass drums.

To provide a foundation for making sense of our findings (as well as to assist those who aren't familiar with acoustic drums), I'll describe the sound of the different drums and their usual musical contexts. The 22-inch, one-headed, blanket-stuffed Ludwig drum produces a low note with lots of attack, few overtones, and not much resonance—basically a tight, dry thump that's often favored for old-school rock, funk, blues, and reggae.

The 20-inch Gretsch, on the other hand, produces a slightly higher note and a rounder, more resonant tone. There's still plenty of attack (thanks to the hole in front, the blanket inside, and the wood beater), but the resonance provided by the front head definitely fills out the sound, resulting in more of a musical note than a quick, dry thump. This more modern sound works in many musical settings, from grunge to jazz.

The 18-inch Slingerland produces the highest note, thanks to its smaller-diameter shell, and is often favored by jazz players. The curious thing, though, is just how huge a little jazz kick can sound when it's recorded. That's partly because 18-inch kick drums are typically tuned fairly open (that is, with

# Crystal Blue Performance

## Crystal Clear Precision


## S200 Features

| | |
|---|---|
| • 20 bit A/D/A | • 15 Incredible Effects |
| • 96dB signal to noise ratio | • Dual Engine Design |
| • Large Custom Display | • Simple User Interface |
| • True Stereo Operation | • 99 User and 99 Factory Presets |


*Each menu in the S200 features simple graphics for effortless operation. The easy-to-use interface doesn't overwhelm you with confusing screens of endless parameters. The S200 lets you get right to the business of making music, not wasting time programming.*


Behind the cool blue exterior of the Digitech S200 lies the impressive clarity of an honest 20 bit A/D D/A conversion, coupled with an ultra-quiet -96dB signal to noise ratio, allowing you to run discreet, crystal clear signal paths. The powerful dual engine processor of the S200 enables you to use any one of the five different effect configurations capable of placing effects in any order. The large custom LCD interface makes any effect or parameter easily accessible. Whether you are in the studio or on the stage, The Digitech S200 has just the right color for any situation.

Other Studio Series 20 bit Processors Include:

**S100**

**Quad 4**

DigiTech STUDIO

A Harman International Company

Digitech • 8760 South Sandy Parkway • Sandy, UT 84070 USA • PH: 801.566.8800 • FAX: 801.566.7005 • www.digitech.com

circle #515 on reader service card


both heads intact and little or no muffling inside), and therefore have lots of resonance and a long decay; but I've also heard the theory that small drums tend to sound bigger simply because the microphone (which is small) can hear more of the drum. At any rate, the sound of the 18-inch drum we recorded was full, round, and very resonant, with a slight "boinginess" and a fair amount of overtones.

Both of Hanes's bass drums were tuned to produce a balanced and versatile tone. Each employed thick batter heads with some form of built-in dampening to cut down on overtones, but neither was outfitted (*in*fitted?) with a pillow or blanket. Although I would characterize both drums as sounding more resonant than dry, the 20-incher—the drum Hanes prefers for his swing and jazz gigs—was the more resonant of the two. Yet both drums produced a well-articulated attack that clarified the thump—overall, a rather modern, rock-type sound.

## HELPER TRACKS

To further our sense of how each mic would perform in real-world applications, we also recorded an adjacent overhead drum track (using a Neumann U 87; see **Fig. 1**) for each pass recorded at Guerrilla. These overhead tracks proved useful during playback for comparing the sound of the bass drum in the room with the sound captured by the kick mics.

Also, Boisen laid down bass lines alongside the drum tracks recorded at Guerrilla so that we could hear how well the kick tracks sat in a mix with the bass—an important consideration. We also tried compressing the tracks, to see how each kick-mic signal would behave in this common processing application. At Boisen's suggestion, we used a UREI LA-4 compressor set to an 8:1 ratio, with only a touch of gain reduction. (The LA-4, a vintage-model, optical compressor with a naturally slow attack and release, is one of Boisen's favorites for kick drum and other low-frequency sources. Bear in mind, however, that a different compressor would likely have yielded different results.)


**FIG. 6: Drums with both heads intact provide maximum resonance but are usually more difficult to mic, especially in a band context. Here, the mic is positioned slightly off center and about three inches back from the head.**

## BOTTOMED OUT

To guard against fatigue, we did additional listening tests on separate days from the recording sessions. At my studio, we listened on Audix 1A monitors; at Guerrilla, we used Event 20/20s (the passive ones). We also made a point of listening to the ADAT tracks recorded at my studio in the control room at Guerrilla Recording. Not surprisingly, the tracks sounded somewhat different depending on the monitors and rooms we used. When played at my studio, the tracks exhibited slightly more bass content.


**FIG. 5: The integral stand clamp on the Shure Beta 52 mic wouldn't work with a squat floor stand, so we had to use a boom.**

To expand the coverage, I enlisted the help of other ears, including those of **EM** Editorial Assistant Rick Weldon and Associate Editor Gino Robair. Weldon plays guitar and bass in several bands, two of which have released CDs that Weldon helped record and mix. Robair, an accomplished drummer, recordist, and producer, owns Rastascan Records, a label with more than 40 titles in its catalog, most of which are devoted to improvisational music. Also, Hanes listened to the tracks he recorded and offered comments.

But before getting into the sonic results of the tests, let's first consider cosmetic and ergonomic issues. The latter are especially important: if a microphone can't easily be attached to a mic stand or is difficult to position once attached, you may soon tire of dealing with it—no matter how good it sounds.

## PHYSICAL EXAM

**AKG D 112.** The D 112 is a curious-looking mic that resembles a small football (or maybe a jumbo egg) stuck in a stirrup. Its wire-mesh grille, bisected by a teal-green, metal protective band, indicates the capsule side. The polar pattern is cardioid.

The shape of the D 112 and the included nylon mic clip make for quick and easy positioning. The clip's nylon threads compromise its durability, though. (I know this because I own a D 112, and

# There are many things a musician can live without. This isn't one of them.


**NEW FEATURES**

- 2 built-in, world-class stereo multi-effect processors with over 300 effects, including COSM-based mic modeling and guitar preamps
- Simultaneous recording on 8 tracks*
- CD recording and archive feature for "first note to final CD" full production capability**
- Slanted backlit display and luminous buttons for low light operation
- 128 virtual tracks allow you to record solo after solo on the same track
- EZ Routing mixer set-up system and automatic configuration makes recording a breeze
- 8 inputs: 6 balanced analog TRS and 2 digital
- 8 outputs: 4 analog and 4 digital
- 16-channel integrated digital mixer with full dynamic automation

*Number of simultaneous recording tracks is dependent on speed of hard drive and recording mode used.

**With optional VS-CDR recording package.

*Specifications and appearance subject to change without notice.*

Introducing the VS-880EX, the latest addition to the acclaimed VS family and the powerful successor to the legendary VS-880 – the world's most popular digital workstation. The VS-880EX takes you to the next level with an array of powerful new features and capabilities that'll make you wonder how you ever lived without them. The VS-880EX. The EX stands for essential.

**Call (800) 386-7575, ext. 596 for your VS-880EX Demo Video ($5.00)**

www.rolandus.com Fax-Back Information: (323) 685-5141, ext. 271 (Doc. #10305)

*Roland Corporation U.S., 7200 Dominion Circle, Los Angeles, CA 90040, (323) 685-5141*

*Roland Canada Music Ltd., 5480 Parkwood Way, Richmond, B.C. V6V 2M4, (604) 270-6626*

Roland

**VS-880EX Digital Studio Workstation**

circle #516 on reader service card

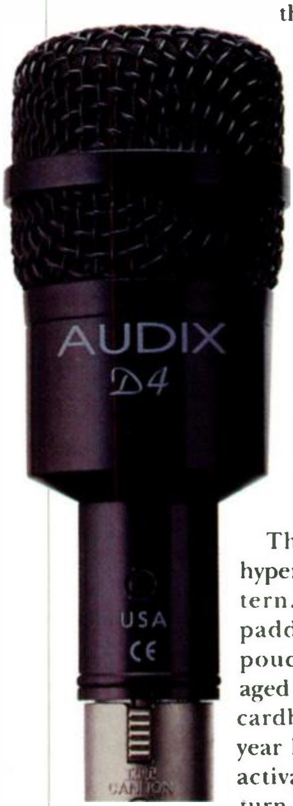

the threads are stripped after a few years of use.)

The D 112 comes in a hard-plastic, foam-lined case with cutouts for the mic and clip. An applications booklet is included, complete with drawings of mic positioning for kick drum, bass-guitar cabinet, double bass, tuba, trombone, and sax. The unit comes with a one-year limited warranty.

**Audio-Technica ATM25.** As kick-drum mics go, the ATM25 is on the small side, but its chunk-o'-steel housing packs considerable heft. Finished in a low-reflecting, gun-metal gray, this little brute sports gold-plated XLR connectors and an integral stand clamp.

Generally, I'm not a fan of integral stand clamps, especially if the design impedes quick setup and positioning. Besides, if the clamps break, they're not as easy to replace as a separate mic clip. But the ATM25's integral clamp is a sturdy, all-steel hunker with an adjustment screw perfectly turned to fit the hand. Thanks to the microphone's small size, I encountered no serious slowdown while attaching the ATM25 to a mic stand. I merely treated the entire assembly as a big wing nut and screwed the mic on. Once it is secure, positioning is a snap. The clamp is easy to remove, too, if you ever need to replace it.

The ATM25 has a hypercardioid polar pattern. It comes in a padded vinyl zipper pouch, which is packaged inside a foam-lined cardboard box. A one-year limited warranty is activated when you return the product registration card.




Audix D4

**Audix D4.** The smallest and lightest of the bunch, the Audix D4 is a cute mic about the size of a saltshaker. Its aluminum body has a sleek, black fusion-coat finish, and its stainless-steel-mesh grille cap seems unusually tough. The polar pattern is hypercardioid, and the XLR connectors are gold-plated.

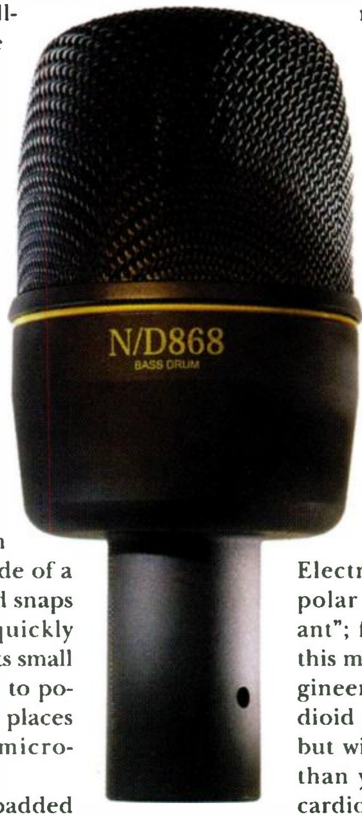
Like the D4 itself, the included mic holder is small and lightweight. Fitted with metal threads, it's made of a sturdy hard plastic and snaps on and off the mic quickly and easily. Thanks to its small size, the D4 is a cinch to position, and it can fit in places where some bigger microphones can't.

The D4 comes in a padded Cordura zipper pouch inside a cardboard box. It carries a two-year warranty against manufacturing defects.

**Beyerdynamic TG-X 50.** The TG-X 50 is a big, side-address mic with a vintage vibe. The body looks like it's made of black Bakelite, but actually it's two metal halves joined together. Rather than a wire-mesh grille, this mic has vents like a radiator's. Two teal-blue stripes and the word "Front" announce the capsule side of the mic. The polar pattern is hypercardioid.

Included with the TG-X 50 is a simple (and quite tough) metal-threaded nylon clip that snaps or slides readily on and off the mic. The clip is easy to position, but it's not particularly snug fitting—I wouldn't happily trust it to hold the mic high in the air over, say, the bell of a tuba, because a hearty blow to the stand could send the mic toppling. For bass drums and other low-to-the-ground applications, though, this shouldn't be a problem.

There is, however, a slight disadvantage to the side-address design of the TG-X 50—at least if you're looking to stick the mic inside the kick drum. Twice, while I was recording the 22-inch Ludwig drum, the TG-X 50 rolled off the foam-and-blanket bed on which I had positioned it inside the shell. I finally had to tape the mic down to get through the track.




**Electro-Voice N/D868**

The TG-X 50 comes in a padded nylon zipper pouch packaged in a cardboard box. The company provides a one-year warranty on parts and labor.

**Electro-Voice N/D868.** The N/D868 is a fat cylinder of a mic with rounded contours and a no-frills look. Its steel body has a flat-black finish and a one-piece, densely woven wire-mesh grille cap. Electro-Voice calls the mic's polar pattern "cardioid variant"; from what I can gather, this means that it has been engineered to provide a full cardioid pickup pattern in front, but with better rear rejection than you typically get with a cardioid pattern.

The microphone holder that comes with the N/D868 is a sturdy, thick plastic affair with metal threads and a knurled metal adjustment knob. It works beautifully, providing quick, secure, and easy positioning.

The N/D868 features gold-plated XLR connectors and comes in an especially nice Cordura zipper pouch complete with a gold-monogrammed logo (and, of course, a cardboard packing box). Also included with each N/D868 is a printout of the frequency response for that particular mic. (Manufacturers often provide only a representative frequency-response plot for each model.)

The warranty is impressive: it guarantees the mic against malfunction from any cause for two years. In addition, the acoustic system contained in the mic is guaranteed for a period of ten years from the date that Electro-Voice discontinues the manufacture of the N/D868.

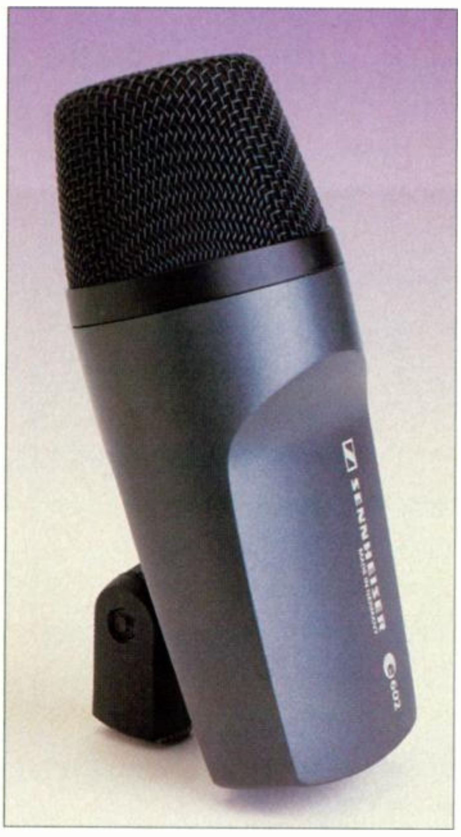
**Sennheiser E602.** The E602 is a large, modern-looking microphone that is shaped something like a shuttle craft from the starship *Enterprise*, but without windows. It has a tapering body with a scalloped top, a low-reflecting gray finish, and a one-piece black wire-mesh grille cap.

TRINITY V3
The world's most dazzling workstation and the wildly acclaimed Z1 synthesizer together in one keyboard. The V3 gives you Trinity's PCM and Z1's sound modeling architectures in one unit. Plus, new combinations and an additional 64 programs. The Z1 board is also available as an option for current Trinity owners.
$100 REBATE!
Get a $100 factory rebate direct from Korg USA with your purchase of a new Trinity V3, V3 Pro or V3 ProX.
See your Korg dealer or www.korg.com for details. Offer good Jan.1–Apr. 15, 1999
KORG
Super sonic.
SEE US AT NAMM BOOTH #406A
© 1999 Korg USA, 316 South Service Road, Melville, NY 11747.
circle #517 on reader service card


We found the E602's integral stand clamp a bit bothersome. The clamp, made of nylon, is connected close to the rear of the mic, throwing off the weight distribution of the whole assembly and making it tricky to get the threads started correctly. The threads are also made of nylon, which increases the likelihood of cross-threading and makes durability a concern. (Even the European-thread adapter is made of nylon.)

On the plus side, there are little hash marks on the clamp that line up with hash marks on the mic, so it's easy to note and repeat the angle at which the E602 is positioned. I can't say I would ever actually use this feature, but I suppose it could come in handy for super-meticulous types.

The E602 employs a cardioid polar pattern and is notable for having the biggest diaphragm of the bunch (one and a half inches in diameter). The mic comes bubble-wrapped inside a cardboard box and has a nylon zipper pouch. A one-year limited warranty is offered.




Sennheiser E602

**Shure Beta 52.** The Beta 52 is a big, heavy, and bulbous mic that looks sort of like a handheld blow dryer. It has sensuous, art-deco curves; a distinctive blue-gray metallic finish; and a large, one-piece wire-mesh grille cap. The polar pattern is supercardioid, and the XLR connectors are gold-plated.

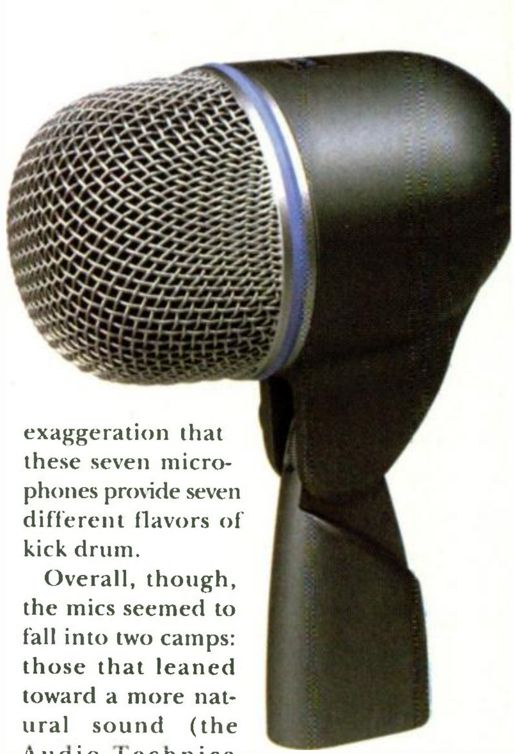
The Beta 52 comes with a unique and very cool-looking integral stand clamp, complete with metal threads. Functionally, though, it leaves a bit to be desired. For one thing, the XLR-connector well is positioned less than one-eighth inch from the thread well. This prevents the use of certain mic stands (including those short Atlas stands that are so convenient for miking kick drums) because the XLR connector on the mic cable gets in the way of the clutch grip. Furthermore, the adjustment screw is small and too rounded off, making it somewhat hard to turn. Also, the size and shape of the mic and clamp together render the whole assembly a bit cumbersome.

Another minor concern is that the wires from the XLR connectors pass outside the stand clamp to reach the mic body. Although they are enclosed within a protective spring, the design would seem to complicate repairs, and the exposure of the spring doesn't inspire confidence in the mic's longevity.

The Beta 52 comes bubble-wrapped inside a cardboard box, along with a large vinyl zipper pouch. The cartridge and the housing are guaranteed for two years, and the transmitter parts are guaranteed for one year.

## NOW HEAR THIS

The first thing we noticed when listening to our recorded tracks was just how different the mics sounded from one another. In a previous test (see "Attack of the Cardioids" in the September 1998 **EM**), Boisen and I had compared eight large-diaphragm condenser mics that spanned a much broader price range—and yet the differences among them were far subtler than those we heard in the kick-drum microphones. It can be said without exaggeration that these seven microphones provide seven different flavors of kick drum.



Shure Beta 52

Overall, though, the mics seemed to fall into two camps: those that leaned toward a more natural sound (the Audio-Technica ATM25, Audix D-4, Beyerdynamic TG-X 50, and Electro-Voice N/D868), and those with a more tailored response (the AKG D 112, Sennheiser E602, and Shure Beta 52). But the delineations weren't always clear, and varying degrees of "natural" and "tailored" sound were evident within both camps.

We also noticed consistent sonic differences between the analog and digital tracks, with the analog tracks sounding fatter, warmer, and punchier in most cases. (But then, it's hardly a secret that drums and bass typically sound better when recorded to analog tape.) This somewhat complicated our findings, just as the other variables did—drums of different sizes, different recording environments, and different ears. But in the end, those same variables brought us to a better understanding of each microphone's sonic disposition.

Tastes, of course, vary—and in music, taste rules. Obviously, when there are seven flavors to choose from, everyone isn't going to pick the same one. (This tendency was amply illustrated by Weldon's consistent preference for the more unusual-sounding bass-drum tones. For one of the tracks, he noted that the sound was "the most like a piece of wood with pieces of metal sticking out of it." Then he added, "That's good.") Therefore, rather than simply declaring a winner, our purpose here is to describe the mics in enough

# PURE CONVERSION.

AUDI/O

***We concentrated first on the quality of conversion with the AUDI/O™ series.*** *Then we added the professional features you need to get the job done right. Nothing fancy, just great sound and ease of use. 24 bits is our standard, as always. ADAT lightpipes provide compatibility, low cost, ease of use, and freedom from ground loops and EMI/RFI. Check them out at your favorite music/pro audio store. Full specifications and suggested list prices available on our website.*

## AD/24
## DA/24

*Eight channels in a standard half rack package. Mix and match octets of conversion. Balanced XLRs (+4/-10) and 117dB S/N ratio make sure your audio is pure. Features word clock, activity/peak LEDs, 48KHz/44.1KHz sample rates.*

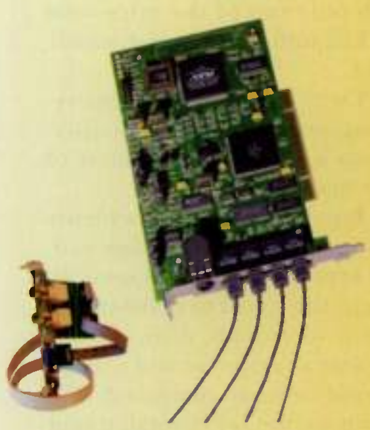
## AD/9624
## DA/9624

*Mastering quality converters run at 96KHz/88.2KHz switchable to 48KHz/44.1KHz. Eight channels made them great for DVD/surround. Can also be used as an AES/EBU interface. Connects to STUDI/O via the S/MUX™ system. Features +4dBu balanced XLR, switchable dither, bit splitting, precision analog level trims, LED indicators for status and signal level.*

## AES/8

*Designed for video post, to connect AES/EBU, SPDIF, and TDIF equipment to your DAW via STUDI/O. Features word/video lock, source switching per channel (AES/TDIF), input sample rate tracking (async. sample rate conversion), pull-up/pull-down locked to NTSC/PAL video, activity LED's on each input and output channel, illuminated buttons, switchable SPDIF/AES output format, full 24-bit signal path (20-bit sample rate conversion).*

## STUDI/O



*Of course, the Award Winning DAW interface which gives you everything you need (pristine 24-bit audio, compatibility, expandability) without everything you don't need (high price). Just about every audio editor around - Mac and PC - is compatible (check our website for up to date compatibility info and application notes), and with the new the S/MUX™ system you can now use your favorite application for 96KHz projects.*

Sonorus, Inc., 111 E. 12th St., NY, NY 10003, USA Phone: +1-212-253-7700 Fax +1-212-253-7701
http://www.sonorus.com info@sonorus.com



SONORUS

SEE US AT NAMM BOOTH #1722

circle #518 on reader service card


detail to give a sense of how each sounds relative to the others, as well as how it handles in the studio. We hope that readers armed with this information can more easily find their way to the sound (or sounds) they're trying to achieve.

## BOTTOMS UP

**AKG D 112.** The D 112 has a signature sound with pleasantly hyped lows and low mids and distinctive-sounding—though not particularly accurate—high mids. This tailoring generally leads to a full, ready-to-mix drum sound with plenty of attack and low end; however, depending on the drum and mic placement, it can also cause the mids to sound a tad scooped out, resulting in a slightly flat or boxy sound. At other times, though, the D 112's low-mid emphasis (around 250 to 300 Hz) results in too much shell tone, sometimes noticeably altering the note of the drum.

Overall, the D 112 was a consistently good performer in all our applications—and it was definitely the most versatile of the three "tailored" mics. It sounded awesome on the 18-inch kick drum ("like a cannon!" I wrote in my notes). On the 20-inch Gretsch, Robair described it as "powerful," with a "strong transient" and "nice-sounding thump."

On the 22-inch Ludwig, the sound was good in both applications, but I preferred the close-miked track, which was tighter and more controlled. Almost everyone else, though, seemed to prefer the increased resonance that emerged when the microphone was positioned farther back.

Of the seven mics we tested, the D 112 exhibited both the least rear rejection and the most off-axis coloration. The snare sound bleeding through it, for example, was always clearly present and distinctly colored, emphasizing the high, thin "crack" of the snare drum. And talk about bleed: on the 20-inch Pearl track, which was recorded in a resonant wood room with high ceilings, the D 112 captured almost enough kit for the one track to serve as the whole drum sound.


Audio-Technica ATM25

The D 112 tracks were quite complementary to Boisen's bass-guitar tracks, offering an immediately usable mix with little or no EQ tweaking. The tracks also fared well with the LA-4 compression, which produced more tone and sustain with no damage to the attack.


AKG D 112

**Audio-Technica ATM25.** The ATM25 produces a solid thud with decent transient response, but on the whole its sound is often lacking in lows and in low mids (depending on the application), resonance, and clarity of attack. The sound therefore tended to be a bit muffled and dull. Also, the ATM25 often seemed farther away from the source than the other mics did.

The ATM25 was not as easy to place in either of the camps ("natural" versus "tailored") as some of the others we tested. Compared with the more tailored mics, for instance, it tended to produce less low end and sometimes a higher note; in contrast to the more natural-sounding mics, it seemed muffled and less immediate, with not as much definition ("click") from the beater.

Overall, the relatively nonresonant ATM25 sounded better on the smaller drums, particularly on the 20-inch Gretsch (which is a fairly resonant drum to begin with). Also, Boisen liked it on the 18-inch Slingerland drum, noting "lots of tone, especially in the midrange" and "lots of good lows," though "not enough attack." Robair, too, noted "lots of very low energy" but felt that the mic "emphasized the wrong frequencies for this application."

Of the two 22-inch Ludwig tracks (close- and distance-miked), I preferred the sound when the ATM25 was positioned closer to the head. The sound seemed almost gated, very thuddy and quick to decay, and with sufficient lows—quite usable for certain kinds of mixes.

When mixed with the bass-guitar tracks, the ATM25 sounded somewhat unfocused or indistinct, without enough of either attack or lows. In Boisen's words, it "just couldn't compete with the bass." The ATM25 didn't do so well with the LA-4 compression, either. (To be fair, the compression we applied to the bass-drum tracks worked well with only two of the mics—the AKG D 112 and the Beyerdynamic TG-X 50.)

The ATM25 provides excellent rejection, especially of high frequencies. Also, it was a bit hotter than most of the other microphones.

**Audix D4.** The D4 captures plenty of attack but rarely enough low end. In fact, I kept wanting to add some 60 or 80 Hz to the signal to make up for the paucity of oomph. Also, despite its fast transient response and reasonably natural tone, as compared with the other mics, the D4's overall sound was rather small and thin, two-dimensional or flat, and not particularly exciting. Altogether, the D4 was our least


favorite of the seven microphones.

The D4 definitely fared best on the smaller drums, especially the double-headed 18-inch Slingerland. Robair and I thought it sounded reasonably good on the 20-inch Gretsch, too.

A tip in Audix's application guide advises, "When using the D4, the closer you get to the beater, the less bass you will have!" Indeed, Audix recommended that, to increase bass response, I position the D4 a bit farther back from the drum in all the applications. However, when I compared the close- and far-miked 22-inch Ludwig tracks, the close-miked track had slightly more low end and fullness and, to my ear, sounded better. (Neither track, however, had as much low end as those recorded with most of the other mics.)

Of course, low end is not always what you want from a kick-drum track, especially in some modern, nonmainstream styles of music. For example, although the sound of the D4 on the Ludwig drum wasn't my favorite, I could imagine it working well in certain rap, hip-hop, or techno productions.

In the mix part of the tests, the D4 blended fairly well with the bass but had—not surprisingly—an overly prominent attack and not enough lows. Again, it worked better with the smaller of the two drums. The LA-4 compression didn't do anything to help the sound.

The D4's off-axis rejection was not very good, either. In fact, it picked up almost as much room sound as the AKG D 112, but without that mic's peculiar high-end coloration.

Although the D4 seemed to be the least well suited to the specific task of recording kick drum, its sound did suggest that it might be a more versatile, all-around instrument microphone than most of the other mics in this comparison.

**Beyerdynamic TG-X 50.** The TG-X 50 has a distinctive (and not always pleasant-sounding) high-mid "presence" boost and sometimes lacks sufficient lows, but otherwise it tended to be the most natural sounding and "honest" of the seven microphones. (This was also apparent when I sang through the mics; the TG-X 50 came closest to representing my voice accurately.) Also, of the bunch, the TG-X 50 had the best transient response ("lots of drive," as Hanes put it) and produced the most dimensional "image." Listening to the TG-X 50 tracks with my eyes closed, I could practically see the drum being hit. And Robair said he felt as if he were being "poked in the eardrum," so present and punchy was the sound.


**Beyerdynamic TG-X 50**

Along with emphasizing attack, the TG-X 50's high-mid boost (which starts at around 1.5 kHz and peaks at 6 kHz) tends to enhance the wood tone of the drum. But it can sound, depending on the application, either a bit "honky" or "cardboardy." The cardboard quality was more evident on the single-headed drum, which of course sounded dry to begin with. The honky sound came out on the 20-inch Gretsch, a drum with a midrange resonance that did not need enhancing.

The TG-X 50 is not a mic for positioning inside the bass drum close to the batter head. The close-miked track sounded boxy and even mildly distorted, with a surfeit of attack ("too pointy," Boisen said) and not much oomph to back it up. Resonance was restored, though, with the mic positioned farther from the head; still, the sound was thin compared to that of the "tailored" mics.

In many applications, the TG-X 50 sounded similar to the Audix D4 (both are in the "natural" camp); however, the TG-X 50 had consistently better transient response, was noticeably more dimensional, and generally sounded bigger.

Interestingly, the TG-X 50 elicited the most mixed—even contradictory—responses from among our testers. What some people liked about it, others didn't. For example, I liked it on the 18-inch jazz kick, where it captured a quite natural sound with remarkable depth of field. Boisen, though, thought it sounded small and too bass-lean in this application. Also, Boisen declared that the sound was "unusable" on the 20-inch Gretsch drum, while Weldon liked it.

Also interesting to note is that initial responses to the TG-X 50 were generally on the negative side—especially when we compared it with the more bass-heavy, "tailored" mics. Yet often, the more we listened to the tracks, the more we liked this mic, due largely to its realism, exceptional presence, and clean, tight sound.

Although it was a clear candidate for some low-end boosting, the TG-X 50 mixed nicely with our bass-guitar tracks. Furthermore, it benefited from the LA-4 compression. (As an aside, Boisen and his studio partner Bart Thurber tried using the TG-X 50 to record bass-guitar cabinet. They were both pleased with the results.)

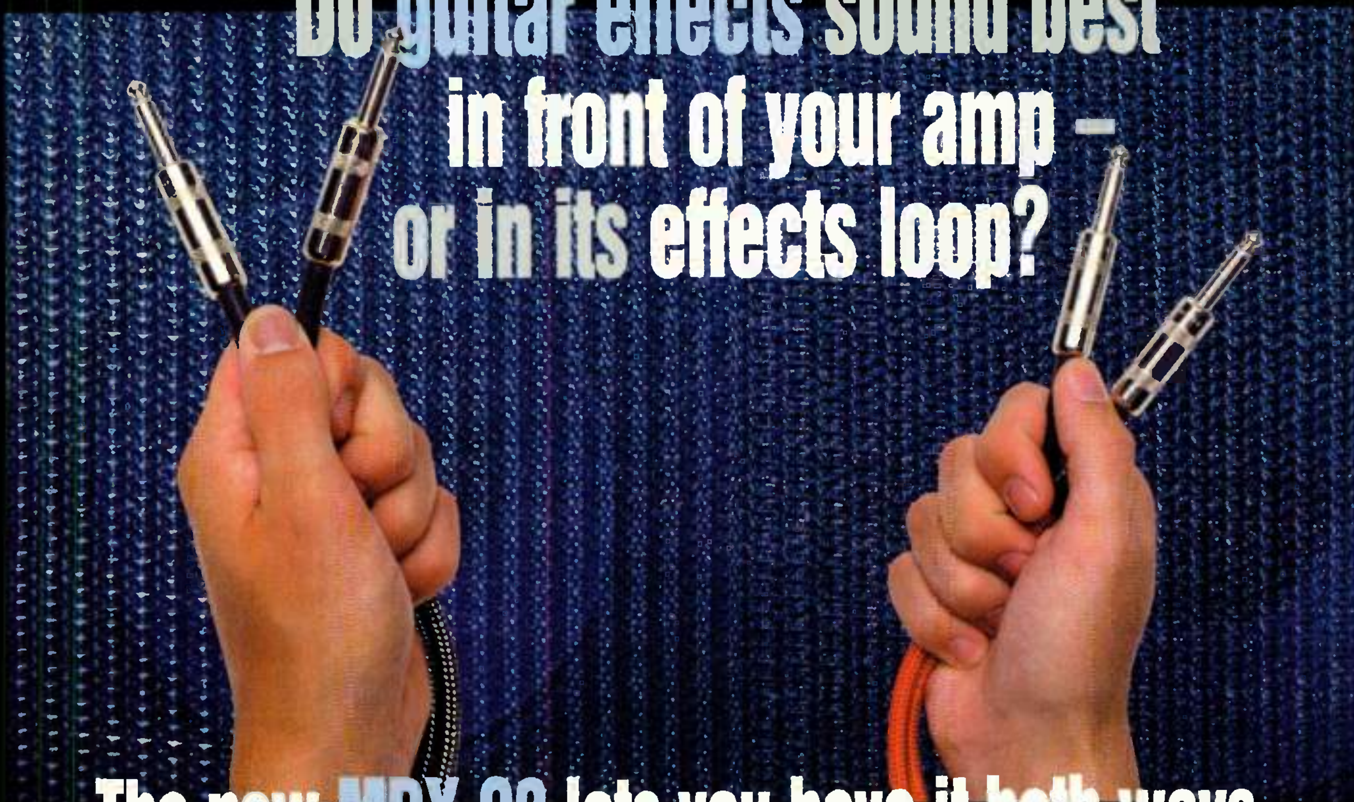
The TG-X 50 was the hottest mic of the bunch; we found that it typically

## Audio Lingo

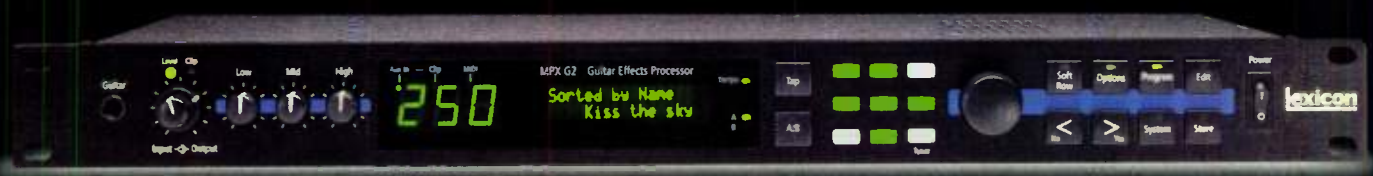
Because the colloquial terms for frequency ranges are so commonly—and inconsistently—bandied about, I have specified the ranges below as they have been referred to in these comparison tests.

| | |
|---|---|
| **Lows** | 20 Hz–80 Hz |
| **Low Mids** | 80 Hz–320 Hz |
| **Mids** | 320 Hz–1.2 kHz |
| **High Mids** | 1.2 kHz–7.5 kHz |
| **Highs** | 7.5 kHz–20 kHz |

# Do guitar effects sound best in front of your amp – or in its effects loop?



## The new MPX G2 lets you have it both ways.




PUT EFFECTS ANYWHERE YOU WANT – in front of your amp, in its effects loop, or both. The MPX G2 is the only guitar processor that lets *you* choose where the effects go - 76 in all, as many as seven at once.

Analog distortion and overdrive, authentic recreations of vintage stomp boxes: Tube Screamer®, Uni Vibe®, Octavia®, Mu-tron III®, Cry Baby®, Phase 90®, Dyna Comp® and more. Studio effects like JamMan™ (20-second looper), Intelligent Pitch Shift, Tap Delay, Chorus, Flange, Rotary Speaker, Lexicon Reverb and Ambience.

Built-in features like a relay bypass and analog noise gate give you pure guitar sounds when you want and let you keep any high-gain amp totally under control. Guitarist-friendly features include dedicated tone controls and a built-in chromatic tuner. When you want to record direct, you can also use the MPX G2 as a direct recording preamp with effects.

Check out the display at your Lexicon Custom Shop dealer and hear it for yourself. Guitar effects sound best through an MPX G2.

*The MPX R1 MIDI Remote Controller gives you stomp-box control of the MPX G2 and your amp for the ultimate touring/studio effects rig.*

lexicon

**Heard In All The Right Places**

A Harman International Company

SEE US AT NAMM BOOTH #645

Lexicon Inc., 3 Oak Park, Bedford, MA 01730-1441 • Tel: 781/280-0300 Fax: 781/280-0490 • Email: info@lexicon.com Web: www.lexicon.com



circle #513 on reader service card


required 6 to 9 dB less gain than most of the other microphones. Off-axis rejection, though natural sounding, wasn't particularly good for a hypercardioid mic.

**Electro-Voice N/D868.** All things considered, the N/D868 was the most consistently pleasing performer of the seven mics we tested, and the one that most of the judges preferred most of the time. In fact, it was one of only two mics that didn't sound inferior in at least one application. (The AKG D 112 was the other.)

Like the Audio-Technica ATM25, the N/D868 falls predominantly into the "natural" camp, although its hyped lows could also qualify the mic's response as "tailored." Certainly its sound is quite different from that of the Beyerdynamic TG-X 50, which was the most all-around natural-sounding mic of the bunch.

My nickname for the N/D868 was Big Thumper. This mic produced a fat, round, warm, and solid thump that was immediately likable but sometimes begged for a modest boost at around 4 kHz to make up for its understated attack and slightly covered sound. Accordingly, Robair described the N/D868's sound as "nicely balanced" and "pleasing," though "a bit dark," and its transient response as "mushier" than the TG-X 50's.

In some ways, the N/D868 resembled the Audio-Technica ATM25 tonally, except that the N/D868 provided more low end and resonance and a lower note overall. Also, the N/D868 sounded less covered and dull.

The N/D868's hearty low end made for a slightly boomy sound in some of the applications (the far-miked 22-inch Ludwig and the double-headed 20-inch Gretsch), but usually the tone, though always resonant, was nicely balanced and the pattern tight. Weldon described the Gretsch track as having a "woody full ring" and noted that the "beater melds right into the shell resonance."

**We were surprised by how different the mics sounded from one another.**

The N/D868 tracks sounded great mixed with the bass-guitar tracks, though again a slight boost at 4 kHz helped define the attack edge. As for the LA-4 compression, it proved unflattering for this mic (as well as for most of the other mics with hyped lows), causing the signal to lose thickness and low end. However, off-axis rejection was excellent—"the least hi-hat bleed of any mic we tested," noted Boisen.

**Sennheiser E602.** The E602 blends chest-thumping lows (in the 50 to 100 Hz range) with smoothly boosted highs and high mids (in the 2 to 12 kHz range) to deliver a huge and very detailed—though far from natural—sound. The result is a highly processed tone that seems perfectly dialed in for arena rock, providing that awesome "kaboom" that makes folks want to raise their Bic lighters and wave them back and forth in salute.

The E602 would probably be a great mic for live-sound engineers to have at a gig where lots of stray bands pass through, as they'd be assured of getting a monstrous kick-drum sound—no matter what lame excuse for a drum set got deposited on the stage. I christened the E602 the Great Equalizer (pun intended) due to its predictable, instantly identifiable response that coaxes from every drum a quite similar, over-the-top sound. The E602's preponderance of lows largely obscures midrange shell tone, minimizing the distinctive voice of each drum in favor of a singular, killer tone. That's not a bad thing if it's consistency and reliability you're after.

In the studio, the E602 was quick to impress, especially when the track was soloed. ("If you find yourself asking, 'Where's the beef?'" quipped Boisen, "here's your answer.") The sound is thick with gut-rumbling lows and sustain, and it packs plenty of punch and clarity without an overabundance of attack. Certainly, if you mix songs for cars that go boom in the night, this is the kick mic for you.

On the other hand, for more mainstream types of mixes, the E602's abundant low end could be too much of a good thing. Even in our simple bass-and-drums mix, the mic's tracks

## Kick Drum Microphone Specifications

| Microphone | Polar Pattern | Diaphragm Size | Frequency Response | Maximum SPL | Price |
|---|---|---|---|---|---|
| **AKG D 112** | cardioid | 1.25" | 20 Hz–17 kHz | unmeasurable | $382 |
| **Audio-Technica ATM25** | hypercardioid | 1" | 30 Hz–15 kHz | not available | $275 |
| **Audix D-4** | hypercardioid | 1" | 38 Hz–19 kHz | 144 dB | $329 |
| **Beyerdynamic TG-X 50** | hypercardioid | 1.25" | 15 Hz–18 kHz (close miked) 40 Hz–16 kHz (@ 1 meter) | 150 dB | $249 |
| **Electro-Voice N/D868** | cardioid variant | 1.25" | 20 Hz–10 kHz | 157 dB | $338 |
| **Sennheiser E602** | cardioid | 1.5" | 20 Hz–16 kHz | 160 dB | $319 |
| **Shure Beta 52** | supercardioid | 1.1" | 20 Hz–10 kHz | 174 dB (@ 1 kHz) | $387.50 |

cakewalk
Pro Audio
24-bit/96kHz
8
Vector Automation
Real-time
MIDI Plug-ins
Sync
AVI
MPEG
QuickTime
Now Shipping
The new version of the leading MIDI and digital audio software
cakewalk
www.cakewalk.com
888-CAKEWALK
All trademarks mentioned are held by their respective owners.


had excessive tone, causing them to compete with and somewhat mask the bass lines. Also, with all that bass going on, the clarity of the attack was compromised.

Not surprisingly, the E602 was something of a mismatch for the more resonant, double-headed kick drums. On the 18-inch Slingerland, for example, the lows were overbearing ("Who needs a bass player?" I scribbled in my notes) and the tone "tanky," like the sound of a rubber kick ball being bounced on concrete. And on my 20-inch Gretsch, the excessive sustain reminded me of a Roland TR-808 kick.

I liked the E602 best on the 22-inch, single-headed Ludwig: whether close- or far-miked, the tracks sounded great. The E602 was also exceptional on the Pearl 20-inch drum recorded to analog tape. In this application, the tone seemed quite natural; in Boisen's words, it had "ample lows" and an "integrated attack."

Like most of the other bass-heavy mics, the E602's sound was not improved by LA-4 compression. On another note, however, this mic provides exceptionally good off-axis rejection—especially considering how much high end it captures.

**Shure Beta 52.** The Beta 52 has a distinctive sound characterized by a big dose of "smacky" attack and ample shell tone but not always enough lows and low mids. The hit is exceptionally immediate and full of impact, and for some styles of music—say, tight funk and disco—this mic would seem ready-made; however, its very "tailored" response resulted in a slightly artificial quality that didn't always float our boats.

A sharp, 12 dB boost around 4 kHz is evidently responsible for the Beta 52's click-heavy attack—an exaggeration that translated better to analog tape than to ADAT, where it usually sounded excessive. On the 22-inch Ludwig tracks, the tone was definitely improved with the mic in the far position. When positioned right up next to the beater, though, the Beta 52 didn't capture nearly enough lows, and the attack sounded rattly and faintly distorted.

The mic's bumped-up mids (starting at around 1 kHz) and high mids didn't always complement the more resonant double-headed drums, either. On the 18-inch Slingerland, for example, the Beta 52 enhanced the drum's "boing" and, in Robair's words, made the track sound "kind of hollow." (Weldon, on the other hand, liked the sound for its quirkiness.) However, on the 20-inch Gretsch, the tone was more usable, with decent lows but still too much attack.

Again, the Beta 52 sounded better on the drums recorded to analog tape, where it captured more lows and had a better-integrated, less clicky attack. Of the two drums we used for our analog tests, the 22-inch fared better and proved the best application for this microphone. In these more favorable applications, the Beta 52 sounded similar to the AKG D 112, but with less low end and more attack. Also, its lows were similar to the Electro-Voice N/D868's—except that there wasn't as much of them.

**The mics seemed to fall into two different camps: natural and tailored.**

The Beta 52's tight sound worked well in the bass-and-drums mix, where it provided the necessary detail and punch. Again, though, we found ourselves reaching for the EQ to boost the lows a bit, particularly on the 20-inch kick track. The LA-4 compression didn't help matters, either, causing a loss of valuable low end.

Off-axis rejection was decent, but the Beta 52's characteristic response was evident in a distinct coloration on the snare drum leakage.

## KICK BOXERS

Well, there you have it: seven different microphones and seven different bass-drum tones. So why, in this age of conformity, would manufacturers produce such different-sounding mics? Well, largely because there's no such thing as a "right" or "wrong" sound—especially when it comes to percussion instruments. What each of these mics embodies, then, is its manufacturer's idea of how a kick-drum mic should sound. And clearly, each has a different idea.

Of course, because there is no right or wrong sound, it's absurd for me to tell you which mic sounds best. After all, each is a well-made, fully professional instrument that is capable of recording pro-quality tracks. However, having gotten familiar with all seven of them, I can definitely say that there are some I would like to own more than others.

My first pick would be the Electro-Voice N/D868. Better yet, I would broaden my palette by acquiring at least two of the seven mics. In that case, my preferences would be the N/D868 and the Beyerdynamic TG-X 50—two microphones that have distinctly different sounds. But again, that's just my opinion.

When asked the same questions—Which mic would you choose if you could afford only one, and which two of the seven would you buy?—both Boisen and Weldon also specified the Electro-Voice N/D868 as their first picks. But they would supplement it with the Sennheiser E602. Interestingly, both gave the same reason for their second choice: they felt they could rely on the E602 to deliver a consistently good sound, no matter what the bass drum itself sounded like.

Robair, on the other hand, chose the Sennheiser E602 as his first pick, and would supplement it with the AKG D 112, were he able to acquire two of the mics. His reasoning was more like mine: both mics sounded great to his ears, but they sounded different enough from each other to cover a lot of ground.

But hey, don't just run out and buy a mic based on *our* preferences. Rather, consider your own particular applications and parameters (types of music, drums, room, and the like), as well as your musical tastes, and then review our detailed descriptions of each microphone. Hopefully, that will steer you in the right direction.

---

**Brian Knave** *is an associate editor at* **EM**. *Thanks to Myles Boisen, Mary Cosola, Carrie Gebstadt, John Hanes, Gino Robair, and Rick Weldon.*

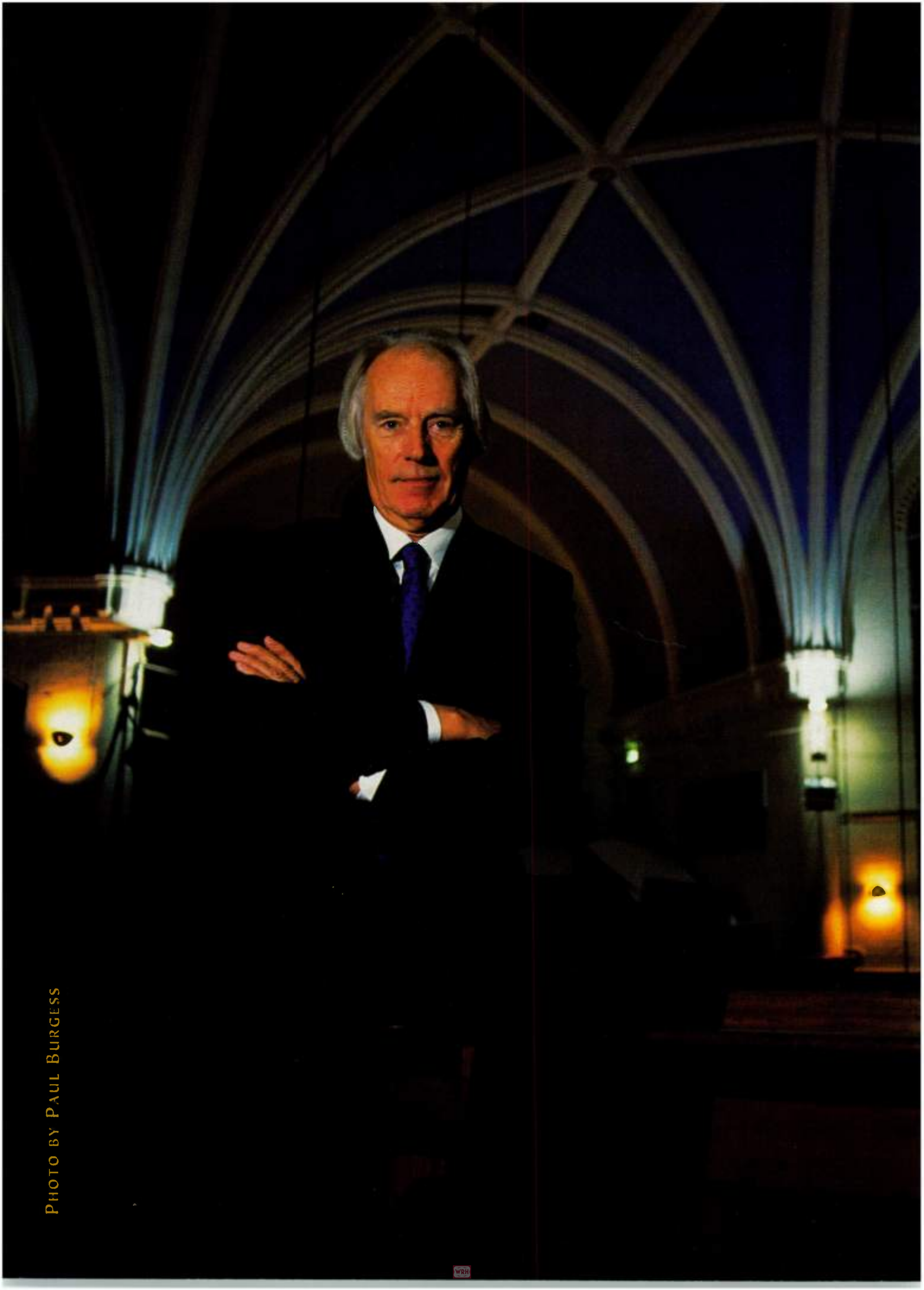


PHOTO BY PAUL BURGESS

By Larry the O

# To Sir with Love

## Conversations with & about Sir George Martin.

To anyone who works with modern sound recording, Sir George Martin is The Man. He has produced 30 singles and 16 albums that reached the top of the British charts and attained similar figures in America and worldwide. He has received countless honors, including five Grammys, knighthood in 1996, and a rare NARAS Trustee's Award. But Martin is best known as the man who signed the Beatles in 1962, when every other label had rejected them. Martin produced every Beatles album except the chaotic *Let It Be*, which bore no producer credit. He has also produced an incredible variety of artists, including Peter Sellers, the Goons, Beyond the Fringe, America, Mahavishnu Orchestra, Seatrain, Jeff Beck, Cilla Black, Elton John, and Peter Ustinov.

Born in England in 1926, Martin started his recording career in 1950. During his years in the studio, not only has he seen the rise of such technical innovations as stereo, multitrack recording, synthesizers, automation, and digital technology in all its myriad faces, but he has pioneered and often invented the now-standard techniques for using these tools. Whether making Beatles records at EMI's Abbey Road studios or working with other artists in his own studios—AIR London, AIR Montserrat (both now history), or AIR Lyndhurst (his current studio)—Martin has always been the master of his sonic domain.

But Martin is more than a great producer and pioneer: he is a true renaissance man of music. As a trained musician (oboe was his original instrument), he brought orchestral instruments to the Beatles records, writing the scores himself and usually conducting them, as well. Martin also performed many of the band's keyboard parts, including the Bach-influenced solo on "In My Life." The avant-garde was just as much his territory, as demonstrated by his score for "I Am the Walrus" and the mad calliope rush of "Being for the Benefit of Mr. Kite." He also composed the scores to films such as *Help* and *Live and Let Die*.


In 1997, he produced Elton John's hit tribute to Princess Diana, "Candle in the Wind '97," which was recorded in just a few hours immediately following Diana's funeral. Finally, Martin decided enough was enough. Suffering from failing hearing, he decided to retire; but first he would produce an album of his own, on which he would work with whomever he wanted. The result was *In My Life*, an album consisting almost entirely of Beatles covers, along with a few of his own compositions, performed by guest artists ranging from (here's that crazy variety again) musicians Celine Dion, John Williams, and Phil Collins to actors Goldie Hawn, Robin Williams, Billy Connolly, and Sean Connery.

Martin's work with the Beatles has been well documented, so I wanted to get a broader view of him and his work. Therefore, I interviewed several others who had worked with him on non-Beatles projects—Andy Kulberg of Seatrain, producer/drummer Narada Michael Walden, Paul Winter of the Winter Consort, violinist Jean Luc Ponty, former Winter Consort member Paul McCandless, and ex–AIR London tape operator Chris Michie (now technical editor of *Mix* magazine, **EM**'s sister publication)—all of whom graciously gave me their insights and memories of the George Martin experience. Their comments are interwoven here with Martin's to give you several takes on some of the same topics.

I spoke with Martin as he was nearing the end of an absolutely brutal week of nonstop interviews and appearances to promote the release of *In My Life*. That evening, at the CD-release party, Martin was feted by MCA executives and fellow megaproducer Quincy Jones. He was given a Gold Record for *In My Life*, as well as the Trustee's Award he had been unable to accept when his wife was going into surgery. Tall, lean, and very British, Martin is unfailingly gracious, even in a state of exhaustion, careful and articulate in his speech, and more humble than one would ever expect for a man of his accomplishments. Although his manner is restrained, his passion for music and sound is clear. Retired he may be; finished working—I don't believe so.

## IN MY LIFE

**One thing I found interesting was that fully half the artists on *In My Life* are actors rather than musicians, and most of those are comedians.**

**Martin:** People have said this. I don't find it surprising at all. I've been in the studios recording since 1950—a long, long time. In 1955 I was given the job of running a label, Parlophone Records, and I was responsible for all the material that went on that label. Because it was a little label that didn't have any big stars on it, I decided to do something a bit different, and I started making spoken word records and musical records with actors and comedians.

The guy I started off with originally was Peter Ustinov. I got to know the Goons very well; they were a cult thing in England, like an early Monty Python. In fact, Monty Python wouldn't have existed without the Goons. Peter Sellers was one of them; Harry Secombe and Spike Milligan were the others. Spike was the guy who did all the writing. I made lots of records with Spike and with Peter, and a whole series of albums with Peter.

Others that I have worked with include Michael Flanders and Donald Swann, the Beyond the Fringe crowd—Peter Cooke, Dudley Moore, Jonathan Miller, and Alan Bennett—Rolf Harris, Bernard Cribbins. The records that I made sold quite well.

I still, obviously, made a great deal of music. In fact, I used to record Matt Munro, who was a very good Frank Sinatra–type singer; a guy named Ron Goodwin, a very good orchestral man who wrote a lot of film music; and Jim Dale, who is now a big Broadway star. Then, in '62, the Beatles came along. When I met with them, they knew about me because they were great fans of Peter Sellers and the Goons, and they knew all the records I'd made. So we had a link already.

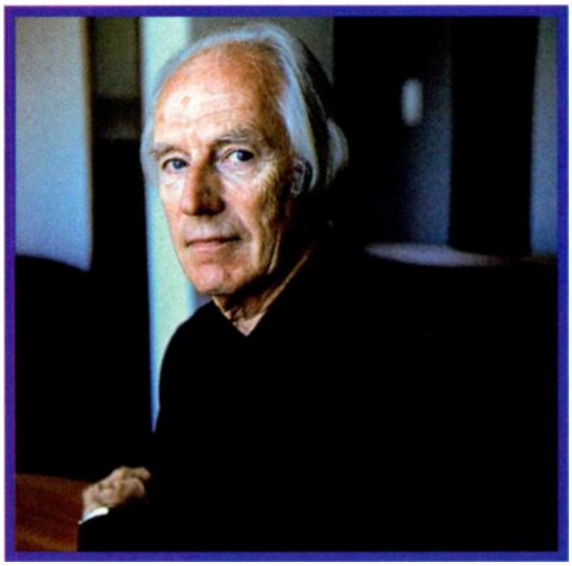
They had a zany sense of humor, also. Without that sense of humor, the Beatles wouldn't have existed, and certainly we wouldn't have hit it off as well as we did. Even after the Beatles, I did covers of certain songs with Peter Sellers; I did "A Hard Day's Night," for example, so it's a kind of tradition. I don't think there's much difference between a performer in music and a performer in spoken word or humor.

**For the songs on *In My Life*, you had written some of the original orchestrations and arrangements, so you had to choose how much to duplicate as well as depart from them.**

**Martin:** Some things were pretty well right. I wanted "I Am the Walrus" on the album. And in putting it on the album, I didn't want to deviate much from where I'd gone with John [Lennon] all those years before, partly because of nostalgia and affection, but also because I think it was the right way to do it. If you did it another way, it wouldn't sit so well.

## DINOSAURS TO SPACE SHUTTLES

**Andy Kulberg of Seatrain commented to me that your first job in recording studios was dropping the weight to turn the flywheel, back when albums were cut directly to disc.**



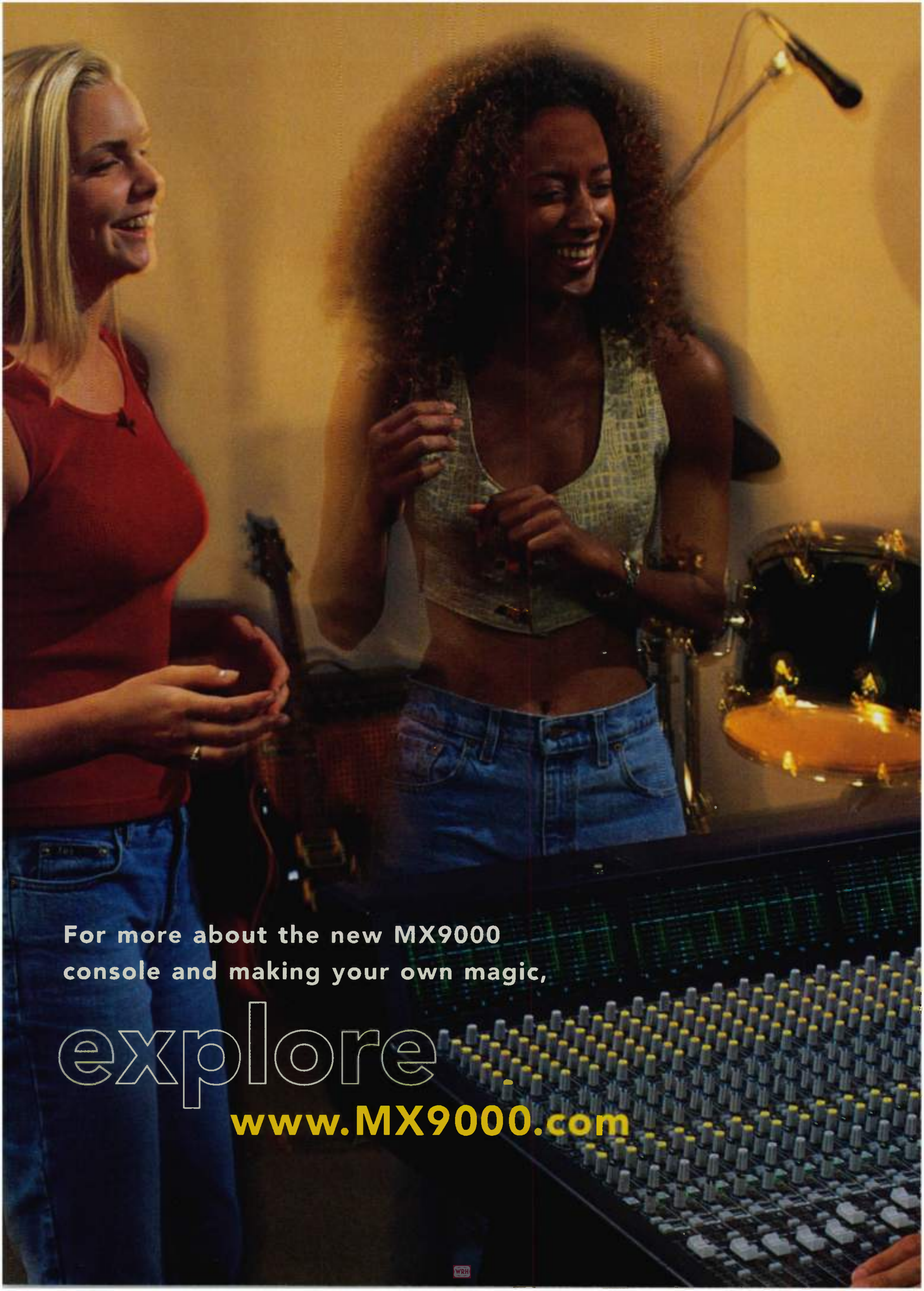
JOHN STODDART

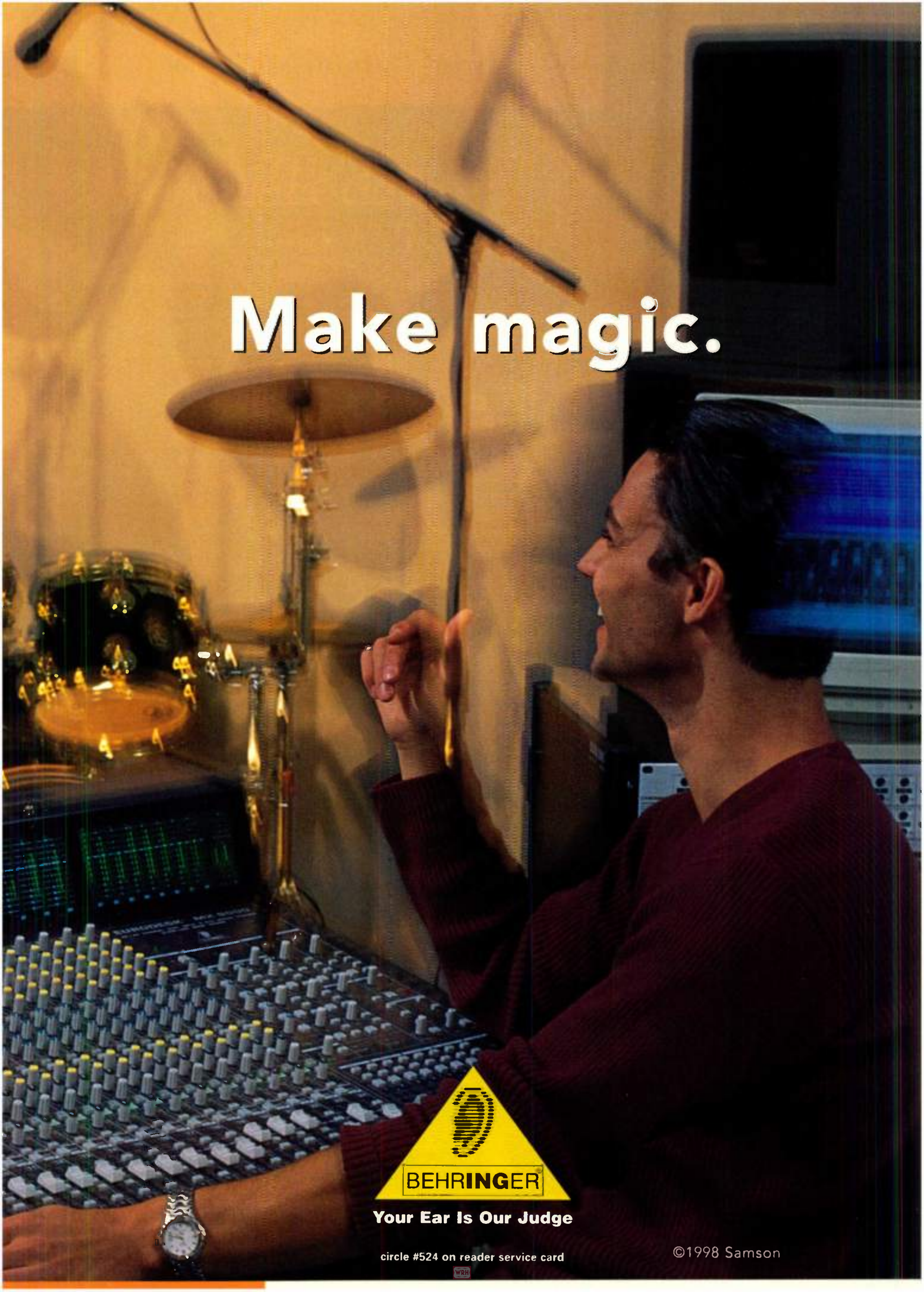
After nearly a half century in the recording industry, Sir George Martin is retiring. His farewell album, *In My Life*, features a variety of musical and nonmusical performers, including Celine Dion, Phil Collins, Jeff Beck, Goldie Hawn, and Robin Williams.

# AKG CONDENSER MICROPHONES TAKE YOU FROM THE STUDIO TO THE STAGE

When you think of high quality microphones, your first consideration should be AKG. Since 1947, AKG has been the leader in condenser microphone technology. Our C1000S, C3000 and C535EB are great for professional and home studio engineers, as well as performing musicians.


- *Perfect for both studio and stage performance*
- *Rugged construction for demanding road work*
- *Shock isolation reduces physically transmitted noises*
- *Competitively priced*


For more about the new MX9000
console and making your own magic,
explore
www.MX9000.com


Make magic.
BEHRINGER
Your Ear Is Our Judge
circle #524 on reader service card
©1998 Samson

# To Sir with Love

to Gershwin, whom he had known.

That album had 18 tracks on it and 17 artists; Elton John had two tracks. It finished up with a version of "Rhapsody in Blue" that was specially scored for Larry Adler way back in the '30s by Richard Russell Bennett, who was a great orchestrator of the time, and I conducted it in Abbey Road's big studio.

When I came to do Elton's songs, he recorded them as two separate items, but I wanted them to flow into one another. We deliberately started off "Our Love Is Here to Stay" with him completely unaccompanied. So it began *[sings and snaps fingers]* "It's very clear...," then the rhythm starts "our love is here to stay," and so on, and then the orchestra comes in. It flowed out of "Someone to Watch Over Me," and I'd written a string link from the end of one to the other so that it became a segue. But when I put the two things together—and I had the benefit of digital editing, so it was quite easy to do—I found that, over the string passage, the entry of the voice was much too brittle. It was not a happy coincidence. The voice jarred when it came in; it was much too bright and perky coming out of the tail end of the previous song.

The way he sang it was *[sings in tempo]* "It's very clear...." It sounded awful, so I thought, "What am I going to do with this if I'm going to work it at all?" Using the computer and digital editing, I was able to stretch the voice without changing the pitch and actually made him sing *[sings first two words slower and more rubato, then in tempo from third word]*, "It's veeryyyyy clear, our love is here to stay...," and it worked beautifully. Then I played it to Elton, and he hadn't any idea he hadn't sung it that way. That's something you couldn't have done in the '60s.

## WORKING WITH SEATRAIN AND THE WINTER CONSORT

**Kulberg:** When Capitol bought Seatrain from A&M, around 1970, we took some of the same material that had appeared on our A&M album and were going to make another debut album, done hipper. Part of the hipper presentation was that we were going to get a big-name producer. We had the crazy idea, of course, of having George Martin produce us.

We recorded a demo and sent it to George to see if he would be interested in producing us. To our complete shock, he was. He and a partner had just opened AIR London, and I think they were looking for something to further their business. We went over and recorded there. It was a beautiful studio, very traditional but well equipped; it was pretty advanced.

**The studio was a more crudely physical place at that time, wasn't it?**

**Kulberg:** Absolutely. George would feed Peter Rowan's guitar, which tended to be a little tinny and thin, through a two-track recorder at a very fast speed and put a little piece of adhesive tape or something on the capstan, and it would make the tape go "ooooi-innnnneeee" when it went the past the capstan. It put a vibrato on it, and it came out with a little delay. He would mix that in, and it really helped fatten up the sound.

That album did pretty well. "Thirteen Questions" was a fairly big hit—we had success with that and could kind of write our own ticket. So, for *Marblehead Messenger* [the second Seatrain album, also produced by Martin], he got a band that was not so demanding of him; we were on our own trip. We were visualizing ourselves as a success, so instead of us going over to London, he came to us. We rented a house in Marblehead, Massachusetts, which is a gorgeous little town.

**Martin:** Seatrain's recordings were very happy recordings. Bennett Glotzer was their manager, and he convinced me to do this series. He also managed a group called the Winter Consort, and we recorded them in tandem.

I knew that doing two albums with two different groups would take quite a bit of time. I had my family to consider: my son Giles, who is the coproducer on *In My Life*, was only two, and I didn't want to be apart from them for a long time. So I said that I would come over but that I wanted to bring my family, and I didn't want to work in New York, either, particularly in July. As the group came from Marblehead, I did a kind of reconnaissance early on, and we rented an empty house on Marblehead Neck. I took my engineer, Bill Price, over with me, and we converted it into a studio. I bought lots of sound-deadening panels, and they fixed them up in the rooms and made a little studio that would work for both groups.

I rented a house nearby and moved in with my family. I got a desk from Rhode Island and a tape machine from somewhere else....We assembled a studio. It was comfortable and very good. Obviously, we didn't have all the finer points

**Robin Williams, George Martin, and Bobby McFerrin consult on "Come Together," which Williams and McFerrin performed on *In My Life*.**


section, and then, of course, the tape would run on into the next section, at which point the mix would totally go to hell. Then they'd set up for that next section and try to match the one before and so on, then edit the 2-track tape together at the same point as the multitrack edit. It was a very painstaking process.

## APOCALYPSE AND WIRED

**There was a story Narada Michael Walden told me about doing the Mahavishnu Orchestra *Apocalypse* album in 1974 at AIR London....**

**Martin:** One of my favorite albums, by the way. I loved "The Smile of the Beyond." It's one of my favorite pieces of music.

**Narada Michael Walden:** To save time, the symphony had to record lines that chased Mahavishnu on "Hymn to Him" ahead of time [before John McLaughlin recorded his parts].

When the rhythm section was recorded, George would conduct and cue me as to where to come in, then cut me, then cue me in again. There was one point where the strings were dragging the time, and I had to drag my time to match them. It was extremely difficult for me, but a piece of cake for him.

**Martin:** When I talked to John about this album, which was very ambitious and complicated, I thought we would do it live, with the Mahavishnu Orchestra and the London Symphony Orchestra—which was conducted by a young musician who is now very famous, Michael Tilson Thomas—playing together in the studio. So we assembled them in the studio, and the work started off with the strings as an introduction, and then the band started, and Michael [Walden]...Michael is not a quiet drummer.

*[General laughter at the gracious understatement.]*

When the drums and guitar started, all hell broke loose. I was standing by Michael Tilson Thomas on the rostrum, and he turned to me and said, "I can't hear anything." And he couldn't. With the noise of the band, you had no idea that the first violin, five feet away, was playing anything. It was that loud. I realized it wasn't going to work that way, so I had quickly to change my mind about how I was going to produce this album.

I booked another studio [at AIR] in the same block of time and put the band in one studio and the orchestra in the big studio and connected them up. We didn't even have the benefit of closed-circuit television in those days; it was just an aural hookup. We did some of the tracks like that, but it was difficult working that way, and it inhibited the band an awful lot.

I then went through the score with John and said, "Look, some of these things will work this way; some will work another way. One way is to record the orchestra first and then get the band to play on top. Another way is to do the band first, and I'll overdub the orchestra. And the third way is to do both live." Therefore, I had to conduct Michael into the ones in which the orchestra recorded first.

It was a very challenging album for me. John's music is complicated anyway; a time signature like 15/16 is quite normal for him. It was one of the most difficult albums I've ever made—and one of the most rewarding—because it's also one of the greatest albums I've ever made.

**Aside from the acoustical problem, what were the other challenges? The complexity of the music?**

**Martin:** Yes, I think so. Knowing where the damn beat lay, many times! This is where Michael excelled; he was on top of that, and he knew exactly where John was.

**What flabbergasted me was that Michael said that was the first time he'd ever been in a recording studio.**

**Martin:** I didn't know that. I was knocked out with what he did; I thought he was a fantastic musician.

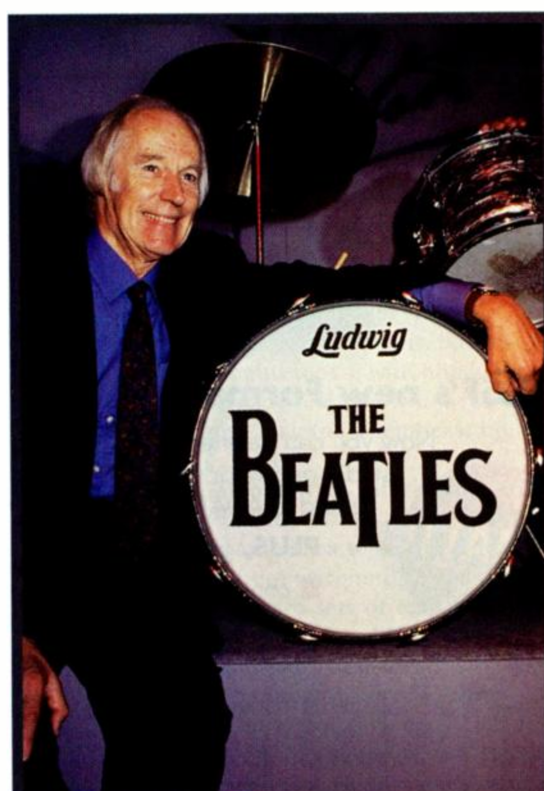
**Walden:** My first time out I worked with George Martin! George was working with people who hadn't even recorded before, like Ralphe [Armstrong, bassist] who was 16 or 17 years old. We had many nights where we asked him about everything: "George, are you sure this is OK?" and he'd say, "It's just great."

He's a musician, an artist; he can arrange the strings and do things that are highly evolved musically, and he's extremely detail oriented. Mahavishnu Orchestra's music was very complex and George could handle it, so that put me at ease. He is a gentle spirit with real musical vision, so you could relax and concentrate on the process of making music and recordings.

**You worked with George Martin again on Jeff Beck's *Wired* album, didn't you?**

**Walden:** Mahavishnu was making the *Inner Worlds* album at Honky Chateau in France. I'd known Jeff through touring with Mahavishnu, and he said, "When you finish making *Inner Worlds,* come over to London." I went back to AIR London, and we worked with George.

On the song "Led Boots," Jeff had given me the signal to start playing drums when the signal had not yet




THEODORE WOOD, RETNA LTD

**In 1995, Martin celebrated the release of the Beatles' *Anthology 1,* for which he remastered and remixed all the tracks.**


come from the [control room] booth to start playing. The recording was all one take, and you can hear the tape machine going "gggglIrrrrr" at the very start. George heard the drum solo at the beginning getting cut off, but he liked it and Jeff liked it, so that went on the record.

## WITH THE BEATLES

**I would be a fool to do this interview without hitting you with a few Beatles questions. *[Martin chuckles.]* You mentioned earlier that you didn't think "I Am the Walrus" could be done as well with any arrangement that strayed from the original. How did that arrangement come about?**

**Martin:** When we did the original recording, John began by teaching me the song in my normal sort of privileged, private performance, where he would sing it in front of me just strumming his acoustic guitar. Of course, if you hear that weird song with just acoustic accompaniment, it's quite different. I was amazed at it and loved it, but I said, "Wow, John, what the hell am I supposed to make of that?"

He said, "I want you to do a score on it."

"Really? What kind of a score?"

"Well, I don't know, the kind of thing you do, strings and horns or something." He was very vague.

I went away and started thinking about it and came up with the score everybody knows. We recorded it in the big studio in Abbey Road [Studio 1]. Unknown to him, I booked a chorus of professional singers who could read music quickly, the kind of people he would normally shy away from. But they were there to do all the little noises, the swoops and the "ha ha has," which were all written into the score.

When John heard it, he hadn't realized how detailed I'd been in my score. He fell around laughing and thought it was great.

**So you scored all of that? I wasn't sure where those ideas had come from.**

**Martin:** It was all in the original score. He gave me suggestions for various things they wanted to do, but ordinarily, we'd have done it with Beatles overdubs. I thought I'd get it done in one fell swoop, and that's the way we did it, and he loved it. Then when we came to mix it, he wanted to go even more far out and had the idea of picking up a radio broadcast and putting it in over the end. By sheer chance, we struck upon someone doing *King Lear* on the radio, and that became part of the mix. We could never remix it like that because that was it.

**It didn't exist outside of the actual mix itself.**

**Martin:** Yeah. To this day, I'm amazed that nobody's ever run up and said, "Hey, you used my voice on that recording; how about paying me something?" Isn't that amazing?

> "HE WAS VERY RESPECTFUL OF ALL THE MUSICIANS INVOLVED AND SUCCEEDED IN BEING CONVINCING WITH HIS SUGGESTIONS WITHOUT EVER BEING AGGRESSIVE."
>
> —JEAN LUC PONTY

**In the days when you were having to do a lot of bouncing, you must have often gotten into situations that were tricky to manage. There were times when you were doing drum overdubs later, which can be tough to pull off.**

**Martin:** The drum overdubs would come about if we felt it needed something heavier—a heavier snare or something like that—or when we wanted tambourine added or whatever. Because of the lack of tracks, we often combined that with a vocal track or a guitar solo or what have you. We'd drop in and out like mad and combine sounds on the same track dangerously, but it was all we could do because I didn't want to go into too much overdubbing. I reckoned bouncing from one 4-track to another 4-track was okay. If you then went to a third generation, it worried me considerably. We did do that sometimes, but I hated doing it.

For example, on the song "She's Leaving Home," I recorded the orchestra on all four tracks: I put violins on two tracks and violas and cellos on the other two tracks because I wanted to be sure I got it so I could really handle it. Then I bounced that down to a stereo pair, which left me with only two tracks for the voices. I knew that I wasn't requiring anything else besides the voices, but I also wanted to double-track them. So I said to John and Paul, "You've got to do this live, both of you singing at the same time." As you know, the song has answering things and there's more echo [reverb] on one voice than the other, that kind of thing.

We put them on two separate mics, so when they got to the *[sings chorus call-and-response parts]* "she is leaving...what did we do with our lives," the "what did we do with our lives" had less echo than the "she is leaving." We got that balance right and got them singing it right the first time round. Then all we had to do was duplicate it—exactly. This is where the Beatles were so good, because they did duplicate it and we got a really good double track, with the same perspective that we needed in the voices, so we didn't have to go to another generation.

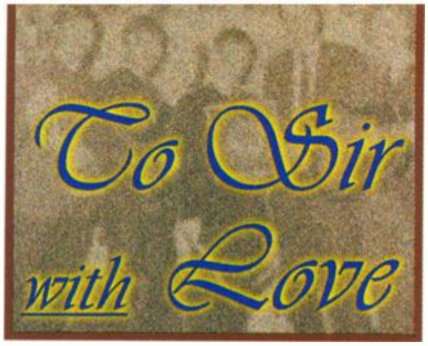
We only had to go to a third generation when we weren't quite sure how to finish the thing or how the arrangement should be topped off, and that's when you would add a sound or something that hadn't been thought of earlier on.

## PRODUCTION STYLE

**When you hear somebody play a raw song, how much are you hearing a finished product or a direction, and how much do you follow your nose?**

**Martin:** It's a gut feeling, really. When you first hear a song, you don't know what's going to come out, and you develop your thoughts about it. You get an idea of the shape of it, and a score evolves in your mind as to what you want.

When I do my scoring, I always try to keep things simple. Every time I put my pen to paper on a score, I ask at each




bar, "Is this necessary? Do you need to have those notes in it? And if you don't, don't use them." Every bar you come to when you're doing an arrangement, you must decide which way to go, how to treat it. You're committing yourself each step along the way, but you have to do that. And sometimes you do it wrongly, but you hope that the law of averages works out okay. I've always felt that, in scoring, simplicity is the keynote, and cleanliness of line. You get that by being very critical of yourself and not pasting in too much stuff.

One problem with 48-track recording today is that you can layer and layer and layer. It's like an artist with a canvas having too many brushes and too much paint. If you keep putting one layer on top of another and saying, "Oh, I really wanted another bit there. Let's put it in and see what happens," then you get a very muddy picture and a very muddy sound.

**I wrote a column recently on the problem of having too many options, and how a lot of great art comes about from artists working against constraints, where it's the application of their creativity against these constraints that pushes them to a level of greatness.**

**Martin:** That's right. It applies to everybody. I remember being most impressed many years ago when I saw a film made for television of Picasso at work. They spent quite a long time on a sequence in which he paints on a ground glass screen with the camera on the other side of it, and you see the paint appearing with the brush strokes. You see him working vaguely and the picture takes shape.

Through time-lapse photography, you get to another day and see the paint changing and him painting over and the picture becoming different. It's a fascinating example of how his mind worked, of the way he would paint. It was a metamorphosis; he would go through different stages.

At one point, I thought, "Oh, that's terrific! Stop there! Don't do any more!" But he went on doing it. Then he came to his version, and I couldn't help reflecting at the end, "I wish he'd stopped because the earlier bit was better than that bit." But he obviously didn't think so. I thought to myself, "If it happens with as great an artist as Picasso, it can happen to us." That's why, always, from that moment on, I've told myself, "Don't go over the top; keep it back."

**And yet, you worked several times with artists like the Beatles and John McLaughlin who push so hard to try and go over the top.**

**Martin:** Yeah, so you had to know where your control lay and how you should guide those people. I probably advised them the way it had to be done. Looking back at the results, I was quite happy.

**Kulberg:** George liked to work on a schedule, which I really respect as a commercial producer. I like to know what's going on; I don't like waiting around for something wild to happen. That's very rock and roll. He's more methodical. You have just so many days to do the tracks, just so many to do the vocals. He would carry around a book of his studio work with the track breakdowns.

He knew a lot from observing, but he allowed the engineer to do all the mechanics, all the touching of tape and the mixing board. He was purely a conduit. He had a very neutral presence when he was working. When you were getting down to the nitty-gritty of cutting tracks or doing vocals, he would listen, but he never let us know what he was listening for. He was a catalyst and, in his own way, definitely a genius. I'm not sure what his exact point of view on things is, other than paying a lot of attention to harmonic detail.

He told me that the first time he ever did a session as a producer, he was assigned by EMI to a jazz group. He was a little uncomfortable with it, went out in the studio and tried to rearrange everything, and they kicked him off the session! As a result, he learned to keep his mouth shut and see what developed, even if he didn't understand it to begin with.

## THE GENTLEMAN PRODUCER

**Kulberg:** He always was an excellent diplomat; that was his forte. He was like right out of *Lawrence of Arabia*—a stoic British diplomat. A gentleman, almost like royalty.

We did a song called "Broken Morning." During the session, he said, "Is that the right note in the bass line?" I was the bass player and had written the song, and one thing that I certainly owned was the bass lines. But I didn't argue with him—I just changed the note to another note, and he seemed satisfied with it. Frankly, I never heard

MICHAEL OCHS ARCHIVES

**Martin and the Fab Four pose during an EMI press conference in 1964. The occasion was a DISC award for "Please Please Me."**



Natural, top quality sound reproduction is the goal of every recording project. And Shure concenser microphones deliver professional results within the realm of a modest budget.

Even the world's top engineers and producers use Shure condensers in the finest recording studios.

To meet various acoustic instrument needs — especially guitar — of any sized recording environment, Shure offers a line of condenser mics that covers instrument recording at all levels: the BG™4.1 for personal studio recording; the SM94 for project studios; the SM81 for professional studios; and the VP88 for stereo recording in any environment, indoors or out.

So to get the biggest sound on any recording format without the biggest expense, there's only one choice — Shure.

For more information, please request your free copy of our full line product catalog and educational booklet, "Microphone Techniques for Studio Recording," by calling 1-800-25-SHURE.



SHURE

THE SOUND OF PROFESSIONALS... WORLDWIDE®

www.shure.com

circle #530 on reader service card

circle #531 on reader service card

SEE US AT NAMM BOOTH #1001

## SOME REMAIN THE SAME

Whether you're working with acoustic or electronic tracks (or a combination of both), some principles of mixing apply across the board. There are six essential rules that I adhere to in *any* mixing situation. No matter how great a mixmaster you think you are, if you ignore these basic principles, your end product will suffer. So, let's run them down quickly.

**1. Be mentally prepared to tackle the mix.** Get yourself in the right frame of mind. Make sure that you're properly rested. This also means taking breaks from the mix periodically; I've found that a breather every two to three hours is sufficient. Avoid interruptions: mixing requires just as much concentration from the engineer as laying down a solo does from the musician. So turn off the phone. Finally, make sure the mood is right. Get a comfortable chair, dim the lights, and fire up the lava lamp. (Okay, the lava lamp is optional.)

**2. Know your client.** What kind of music are you working on? If it's a band project, has the band previously released a record that you can listen to? What are the producer's goals? You can completely alter the sound of a record in the mix, so you need to know what direction to go before you start working.

**3. Be familiar with the monitors.** If you're not working in your own studio, or if the producer brings unfamiliar speakers, give yourself a crash course in monitoring on the equipment. Pop in a CD that you are *thoroughly* familiar with; listen for exaggerated or muffled frequencies, paying particular attention to the low- and high-end content. Make mental (or even written) notes. Once you feel that you know the speakers' response, start working on the mix immediately.

**4. Monitor at low levels.** There are three reasons for keeping the monitors low while you mix. First of all, you'll avoid a case of listening fatigue (not to mention other health-related problems associated with loud music). Secondly, just about any mix sounds good when it's cranked up; the stellar mixes are the ones that sound good both loud and soft. Third, with the speakers at loud volumes you won't catch level problems. I was mixing a radio promo recently and thought I had a great mix going, but I was completely crushed when I lowered the monitors to a normal level—I couldn't even hear the announcer's voice.

**5. Reference your mix to similar commercial mixes.** Listen to mixes that are similar in style to what you're working on. Again, if the band has other material you can use, great. If your console has a 2-track input for a CD player, use it. This way you can A/B between your own mix and the one you're striving to emulate. This step obviously isn't appropriate for those projects you have no intention of patterning after someone else's work, but it's often useful when producing commercial music, especially when you have to please a record label executive.

**6. Reference the mix on a variety of systems.** This is the most important point to remember: a well-balanced mix will sound *good* on poor monitors and *great* on good monitors. Check what you're mixing through several systems of varying size and quality. Before I complete a mix, I've listened to it on my studio monitors, a pair of headphones, a boom box, and my car stereo. If I really want to go crazy, I'll burn a CD and check it out on whatever system I can find. (I've actually done referencing in the electronics department at Sears.)

If you keep these six points in mind while you work, your mixes will improve 100 percent—I guarantee it. (For more information on basic mix principles, particularly with acoustic instruments, refer to "In Your Face Mixing" in the May 1998 EM.)

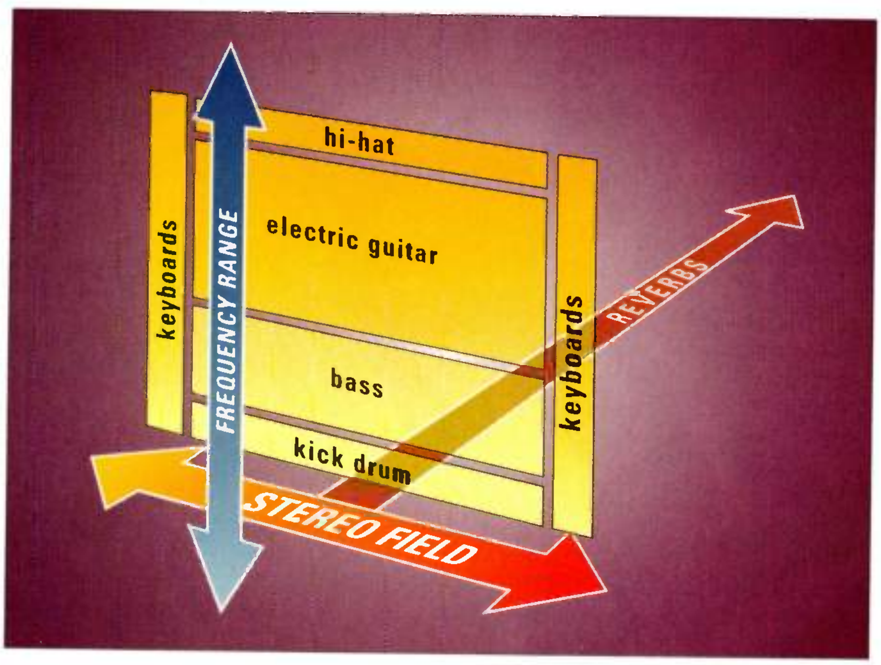



**FIG. 1: By conceptualizing the mix as a three-dimensional stage where the vertical axis represents frequency response, you can graph the soundstage from the front or above and see the frequency/pan position or volume/pan position, respectively. Make sure no two components are in the exact same place in either axis.**

## ROUGH AND READY

If you're mixing on a computer, you have the advantage of building your mix during the recording process so that when it finally comes time to print to 2-track, only minor adjustments will be needed. I once produced a project when we didn't even have a dedicated mix day; instead, I took an hour to automate the vocal track and then printed the song to DAT.

Even if you're not working with a computer-based system, start getting some ideas together as you record. One of the best bits of advice I can impart is to periodically record rough mixes during tracking. All too often, after listening to the same song for weeks or months, we lose the fresh perspective that we had at the beginning of a

# PROFESSIONAL FEATURES, POWERFUL PERFORMANCE, AFFORDABLY PRICED.

## THE AKG WMS 300 FREQUENCY AGILE UHF WIRELESS MICROPHONE SYSTEM.

# WMS 300


Now you can enjoy AKG's legendary microphones on a no-nonsense wireless system. Superb acoustic performance coupled with trouble-free UHF transmission is only the beginning. Each WMS 300 system features 16 user selectable frequencies so interference free operation is only a click away. Need multiple channels? No problem. 16 or more WMS 300 systems can be operated simultaneously without intermodulation distortion. Need to mix systems? AKG's finest handheld, headset, lavalier and instrument microphones are available with the WMS 300.

- 16 intermodulation free user selectable frequencies
- True diversity receiver
- Modular handheld transmitter with interchangeable capsules
- Rugged all-metal bodypack
- 12 hours operation from 3, AA batteries
- AKG's finest microphone capsules

**AFTER BECOMING A LEGEND IN THE STUDIO, WE'VE HIT THE ROAD.**

AKG Acoustics U.S., 1449 Donelson Pike, Nashville, TN 37217,
phone: 615-360-0499, fax: 615-360-0275.
AKG Acoustics G.m.b.H. Vienna/Austria. http://www.akg-acoustics.com

AKG

A Harman International Company

ADV 547/1/E

circle #532 on reader service card


project. Rough mixes are the perfect way to recall that lost perspective. Check in with them often to find out what your ideas were weeks ago.

## WHAT GOES WHERE?

When working entirely with sequenced MIDI parts, you have the option of mixing tracks virtually without ever recording them to a multitrack. The benefits of mixing directly from the sound modules are obvious. For one thing, your signal path is shorter, so your chances of collecting sonic garbage drop substantially. In addition, sticking with MIDI tracks leaves more audio tracks available. But most important, it means you're not committed to anything—you can edit and automate synth sounds and sequences in ways that would be difficult to do with audio tracks, and you can do it at the last minute if necessary. (If you're performing virtual mixing with a digital audio workstation, make sure your system has the ability to mix live inputs and that you have enough I/O available.)

However, many professional engineers like to print MIDI sequences to multitrack audio media (especially analog two-inch tape). Sometimes it's unrealistic for the personal-studio owner to go this route, but it has several advantages. In a small studio with limited resources, for example, recording to audio multitrack allows you to apply outboard effects to individual tracks and submixes, freeing up your limited supply of effects processors for reuse at mixdown. In addition, submixing to tape or disk can simplify your final mixdown process. If you print to analog tape, of course, the tape recorder operates as a signal processor in the sense that it can add a desirable sonic quality.

Whether you mix virtually or print your sequencer parts to tape, it's generally better to clean up the signals at the sound modules rather than at the mixer. Selective filtering and compression can be used to remove unwanted frequencies or dynamics before a sound leaves the module. However, I usually try to save elaborate effects processing for the mixing environment, where I can employ dedicated units (unless I'm mixing on a DAW, where DSP is a precious commodity—in which case I might do *some* processing at the modules and some at the mixer). It's a balancing act, but it's better to have too many options than too few.

If your mixer doesn't offer dynamic automation, you can use MIDI Control Change 7 (Volume) and 10 (Pan) messages to automate your sequenced tracks. The drawback to this is that a mixer channel will remain open even when no signal is present, which is not the case when you manually ride the faders. With an analog mixer, you'll get some hiss; to fix this, you can use noise gates or expanders at the channel inserts.

In general, it's good practice to maintain as many dedicated instrument channels as possible. Granted, sometimes you may have no choice but to submix several parts to a stereo output, for instance. Just avoid unnecessary submixing; the more signals you have to work with at mixdown, the better.

## ACOUSTICS AND ELECTRONICS

There are three basic kinds of electronic sounds: emulations or samples of acoustic or electric instruments; completely artificial sounds; and emulations or samples of real-world, nonmusical sounds (a dog barking, for example). The mix engineer needs to approach each type differently.

When working with acoustic and electric instrument sounds, the goal is usually to replicate an accurate image of each instrument, positioning it in a realistic place on a virtual soundstage and making sure its frequency content is similar to what it would be in the real world. You'll then give the recorded instrument a volume and depth on that stage by using level control, reverb, and sometimes other processing, such as delay. (This is not a hard-and-fast rule, obviously; there are no rules in the creative arts.)

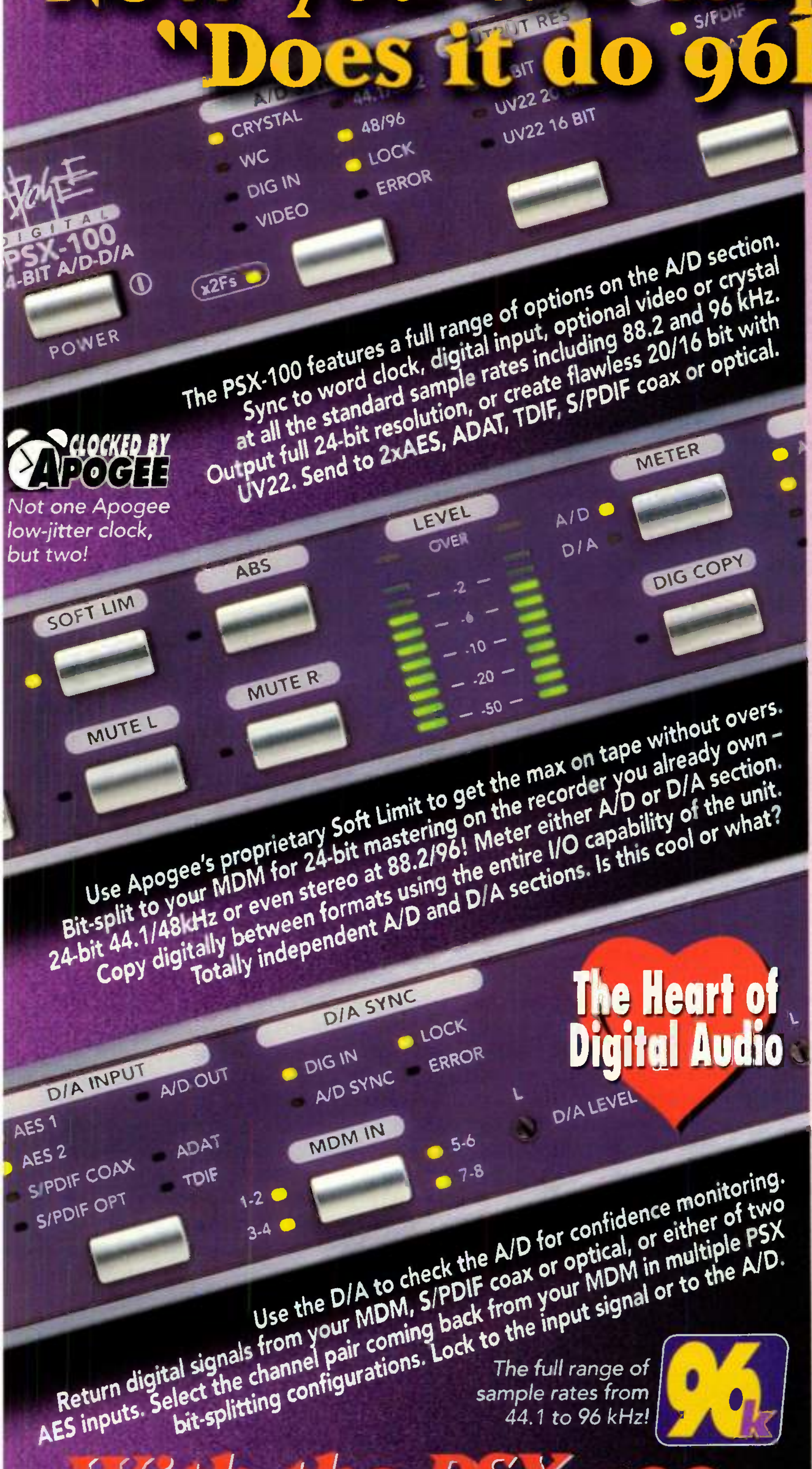
The same philosophy usually holds true if the "acoustic" or "electric" instrument happens to be a sample or synth patch. For example, even though the Roland JV-1080's grand-piano patches are electronic samples, I'd probably still EQ them like real pianos and put them in realistic positions in the stereo image, unless I was trying to achieve a weird result. However, the methods you use ultimately depend on the style of music you're producing: for alternative and urban styles, perhaps the piano would need to be equalized like a guitar and spread across the entire stage.

When working with completely synthetic sounds, a different set of rules applies. Trying to place these instruments in a realistic spot on a "stage"—an acoustic environment where they wouldn't normally be heard—is pointless. In addition, there are no real-world templates of synthetic sounds on which to base EQ settings; I mean, what is a Telefunken or a Space Warp Pad supposed to sound like, anyway? The same is true of nonmusical samples (unless, of course, one of your band members is really a barking dog). The only exception here is when you are creating music for picture and want to position the effects to match the action.

Working with synthetic sounds essentially gives you carte blanche to create exciting mixes with sounds coming

**FIG. 2: In this example, the kick drum and one drum loop are right in the center of the mix, as in a live band. Because the synth bass has a heavy low-frequency content, I have panned it in the center, which also helps lock it in with the kick. The lead vocal and guitar solo are in the center and set hot. They don't occur at the same time, and they have different spectral content than the kick drum, drum loop, and bass, so each instrument will be distinct.**

# Now you can stop asking "Does it do 96kHz?"




**With the PSX-100, the answer is "yes."**

APOGEE ELECTRONICS has a reputation for producing the very best sounding digital conversion products available. You might also think of us as being quick off the mark. But in fact, we think very carefully before we release a new product.

That's how it was when we came to design our first 24-bit, 96kHz converter system: the **PSX-100**.

Anyone can make 24 bits dance up and down 96,000 times a second. The question is, does it sound any better? *In many cases, the answer is no* – and to our engineers, there was no point if the sound wasn't significantly improved. So clock circuitry had to be even more rigorously designed to minimize jitter. Analog components required special characteristics to realize the performance we specified. And much more.

We wanted to give our first 96kHz system all the features needed to make it truly indispensable. AES, S/PDIF, ADAT and TDIF I/O for highest flexibility. Bit-splitting for 24-bit, 96kHz mastering to 16-bit, 48kHz MDMs. UV22® for flawless 16- and 20-bit output. Soft Limit® for maximum level without overs. Digital copy, confidence and monitoring modes. Sync to word clock and optional video.

The result: a total stereo conversion solution. ***Fully independent A/D and D/A converters in a single 1U package.***

People kept asking us, "Does it do 96kHz?" and we said, "When we're ready, it will." Now, with the **PSX-100**, we're ready. Are you?

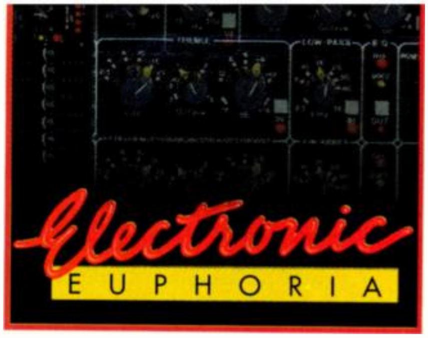
APOGEE ELECTRONICS
www.apogeedigital.com


Apogee Electronics Corporation,
3145 Donald Douglas Loop South,
Santa Monica, California 90405, USA.
Tel: +1 310/915-1000 Fax: +1 310/391-6262.
Email: info@apogeedigital.com.




SEE US AT NAMM BOOTH #W241

circle #533 on reader service card




from all over the stereo image and frequency spectrum. And using creative dynamics control and multi-effects processing, you can mold those sounds into practically anything you want.

## SPATIAL PLACEMENT

You have much more creative liberty with a mix of electronic instruments than you do with a mix of acoustic ones. I like to create a natural-sounding blend of all the elements. Artists such as Beck, Nine Inch Nails, Jane's Addiction, and Alanis Morrisette often employ contrasting timbres that don't blend smoothly—but for a lot of music, a smooth blend is preferable.

In most cases, instruments and sounds should not compete with one another either spatially or spectrally. You should be able to hear every part of a mix and immediately identify which instrument is which. To do this, I try to conceptualize the mix as a three-dimensional stage (see **Fig. 1**). Panning instruments moves them across the width of the stage; altering their level and adding reverb or other delay effects determines how far back they are. The vertical axis represents frequency response (for example, cymbals would be toward the top of the stage, with the kick drum sitting near the bottom). This way you can graph the stage from either the front or above and see the two most important relationships of a mix: frequency/pan position and volume/pan position.

The goal is to make sure that no two components are centered at the exact same place in either graph. I don't mean that things can't overlap—the lower keys of a piano will inevitably be situated in the same area of the frequency spectrum as the bass—but elements shouldn't blatantly sit on top of each other. This is what causes a mix to become cluttered and muddy sounding. A clear mix is achieved through careful planning and adjustment of level, pan, and EQ.

## PANNING AND PLACEMENT

Many people don't realize just how much EQ and level can affect the placement of an instrument in the stereo field. To check this out yourself, try the following experiment. Put the faders of your bass and kick drum tracks up with both channels panned to center. Boost them both by 15 dB at 200 Hz and turn up your monitors. You'll notice that it becomes difficult to distinguish the hit of the kick drum. Now pan the bass to nine o'clock and the kick drum to three o'clock. When you do, the kick drum hit returns. Finally, pan them both back to center and pull down the level of the bass track; you can again hear the kick drum better when the bass is lower.

**Proper EQ means more than just getting a great sound from a track.**

This experiment illustrates how instruments can occupy the same frequency ranges, provided they aren't at the same spatial position in a mix (and vice versa). I'll discuss specific EQ applications below, but this is important to keep in mind when panning tracks.

In general, instruments that comprise the rhythm section are kept toward the center of the mix (see **Fig. 2**). Specifically, drum parts, bass parts, certain pianos, and loops should be spread no further than ten and two o'clock. In fact, try putting monotonous loops in mono; this opens up the horizontal axis for the supporting characters (guitars, piano, strings, and so on). Whatever you do, don't pan your drum tracks across the entire stereo image: have you ever seen an acoustic drum kit with toms that run from stage left to stage right? Keep the kit in the middle.

As a rule of thumb, any part that has a heavy low-frequency content should be situated toward the center of the mix. Simply panning the two signals is a prescription for trouble. True, Beatles engineer Geoff Emerick split the drums and bass to opposite channels on certain Beatles tunes. However, his drum and bass panning was almost certainly done out of necessity for bouncing tracks to overcome track limitations.

Slightly outside the rhythm section lie the supporting instruments: pianos, strings, guitars, and horns. This is also where you might want to place certain background vocals and percussion.

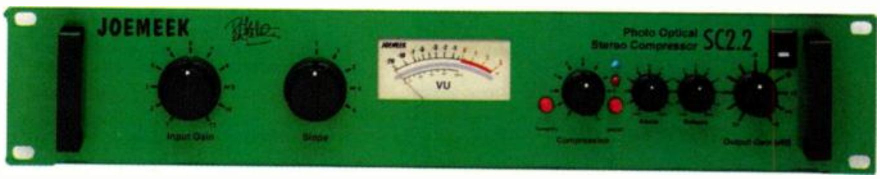
Lead vocals are generally put right up the middle; anywhere else makes them just distracting to listen to. Although instrument solos are usually panned dead-center, I prefer to spread them slightly to either side. Often, a solo will be played along with the lead vocal, and if both are in the center they will be competing with each other. (Many solo instruments have a frequency range similar to that of the human voice.)

Finally, the outside edges of the mix are usually reserved for effects returns (particularly reverb), for certain types of percussion, and for high-frequency background vocals (à la the Bee Gees). Be careful when placing sounds completely in one channel or the other, especially if you're also processing them with multi-effects; in these instances, delays and reverbs that are panned opposite the source sound can cause phase cancellation.

Auto-panning synth patches should be addressed with caution. At what spectral position do these sounds start their journey, and where do they end up? This path must be clear of other sounds in the same frequency range; otherwise, dropouts will occur. Once you have a clear idea of where you want to place everything, it's time to make sure that instruments sitting in similar places aren't competing for room in the frequency spectrum.

## ISSUES OF TIMBRE

Proper EQ means more than just getting a great sound from a track; it's about eliminating congestion in the mix. I want to stress the importance of




**Light compression with a unit like the Joemeek SC2.2 is often best for electronic instruments.**


subtractive EQ. In general, your mix will benefit more from cutting than from boosting. To tweak a sound with a parametric EQ, I usually start by doing the opposite of what I was taught in school: I turn the EQ gain *down* all the way and sweep the frequency knob, so I don't even think about boosting anything unless I really have to. However, if you cut enough frequencies, you'll probably need to make up gain at some point. Fortunately, many digital mixing consoles and DAWs provide a "gain makeup" capability as part of the EQ section.

With electronic replications of acoustic instruments, your best bet is probably to retain the authenticity of that instrument's natural sound. This does not mean that the samples don't need equalizing: although they are supposed to be accurate, pristine recordings of acoustic instruments, many are flawed. Basically, you want to eliminate frequencies that aren't needed. What is the instrument's primary range? In other words, which frequencies are needed to get it to cut through a mix, when that is desired, or to keep it back in the mix when it is supposed to be part of a pad? Once you determine the range, you can whittle it down to the necessary frequencies.

Two of the most important sounds in a mix are the kick drum and the bass part, and there should be a synergy to their relationship. These two tracks constitute most of the low-frequency energy in a mix. You need to decide which of these tracks will be the primary source of low end. For a traditional mix, I usually opt for bass, simply because there is more motion to it, and spotlighting it makes the low end more interesting.


**Equalizers such as this Drawmer 1961 are useful for timbrally separating instruments that are in the same range.**

On a bass part, I find that rolling off frequencies below 50 Hz is a good start. Boosting frequencies in the 300 Hz to 1.5 kHz range (admittedly a large range) will increase the track's clarity, and pulling them out will round out the low end. Once I have the bass sound, I work the kick drum in as support, accenting mid frequencies (1 kHz) and boosting a little around 80 Hz (3 to 6 dB, with a narrow bandwidth).

On the other hand, if you're producing urban music, the kick drum sample or synth patch should be the more pronounced low-frequency element. Because many hip-hop beats are derived from premixed loops, you'll want to boost the track by about 3 dB at 120 Hz. Loops also have little high-end clarity, so you may have to roll off upper frequencies with a shelving filter (usually above 7 kHz). A mid-range boost might also be in order. Most other electronic percussion instruments are fine with little or no EQ; if EQ is needed, it's usually a boost at 7 kHz or higher.

Certain synth pads—especially organ sounds—tend to be heavy on the low-end content. You'll probably find that, although they sound really fat by themselves, they just don't sit well with the rhythm instruments. Try rolling off frequencies below 300 Hz or boosting the track in the 2 kHz to 3 kHz band and lowering the fader level.

Digital pianos can often present the same problem in a mix, especially when the lower half of the keyboard is being used. As long as the piano isn't the only instrument playing, my advice is to roll off the low end. I recently used snapshots to automate the EQ of a piano track in a rock song. The piano started the song out and needed to sound full, but when the rest of the band came in, it totally clashed with the bass. So I simply set up two snapshots—the second one with a highpass filter engaged—and performed the change on the fly.

Electronic strings should accent the high-mid and upper frequency ranges, so a little boost might be needed somewhere above 5 kHz. Try rolling the low frequencies off below 500 Hz.

One very useful application of EQ is reducing hiss from sound modules. Although expanders and gates are an option, you'll find that rolling off frequencies above 10 kHz is sometimes a better approach. Keep in mind that this will work only on synth outputs handling signals that have no frequency content above 10 kHz (such as drum tracks, bass parts, and guitar parts).

Remember, every change you make to one track—no matter how small—will affect the other tracks as well. A tweak to

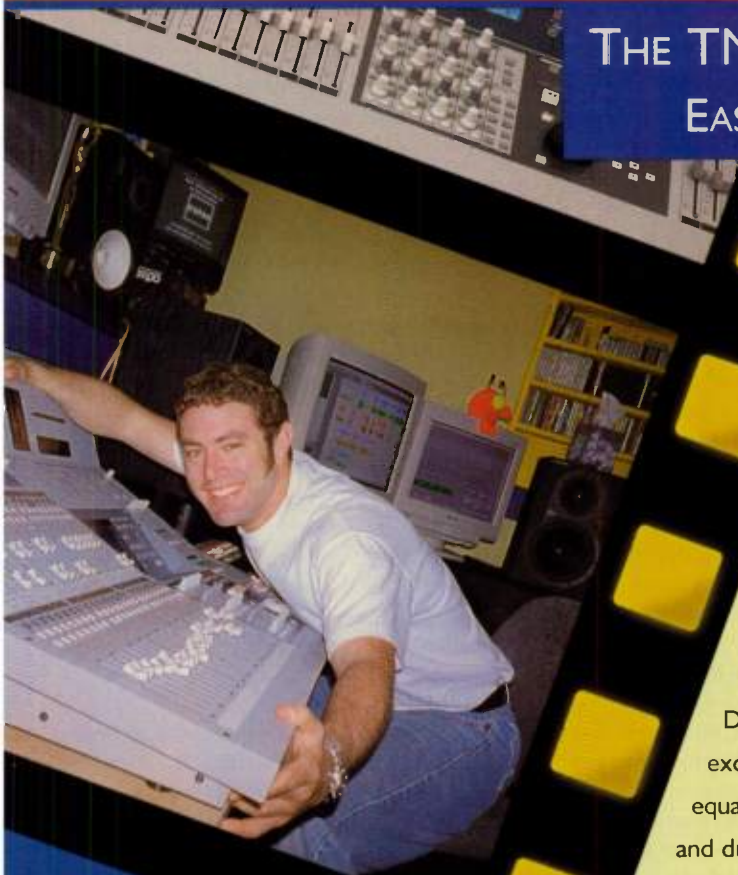
## BUILDING A MIX IN TEN STEPS

Many methods can work for organizing the mixing process. This straightforward ten-step process works well for me. Obviously, there can be much more to building a mix, but sometimes it pays to keep things simple.

1. Set pan positions.
2. Set levels to build a rough mix.
3. EQ each track in context with the others, soloing tracks to isolate problems.
4. Bring all the faders down.
5. Bring your most important track up to 80 percent volume.
6. If needed, process this same track with multi-effects.
7. Bring in supporting instruments, adding effects as needed.
8. Patch dynamics processors across tracks that require them.
9. Make EQ tweaks and check the mix in mono.
10. Adjust the levels of the effects returns.

Sound Makes The Picture. You Make The Sound. TASCAM Makes The Tools.

# The TM-D8000 Digital Mixer
## Easy To Learn, Easy To Use



Scott Strain likes control. He also likes to keep it simple. That's why he chose the TASCAM TM-D8000 Digital Mixer. Just minutes after you plug it in you'll be blazing through Foley, ADR, music production and stem mixing without having to consult the manual at every turn. The TM-D8000 is designed to work with TASCAM's legendary DA-88, DA-98 and MMR Series Digital Multitracks. And with TASCAM's exclusive OpenPath™ Technology, it works equally well with other popular digital multitracks and dubbers. Configured with 16 analog inputs (switchable to digital) and 24 digital multitrack I/Os you get 40 available digital tracks at mixdown.

Screen menus are easy to navigate, and scene recall and mix automation functions including EQ and dynamics libraries are at your fingertips. You have instant access to full parametric 4-band EQ, panning, aux and effects send/returns, and other mixer functions. Plus, you get total control of video decks, digital multitracks, DATs, and other digital gear via built-in MMC and Sony P-2 protocols. At final mix, your options range from stereo to 5.1 Surround Sound, a necessity in today's post environment. Check out the intuitive, foolproof TM-D8000. It proves once and for all that a great digital mixer shouldn't complicate your life. Get full details from TASCAM Faxback. Call 800-827-2268 and request document #8000.

"I like the TM-D8000 because it looks and feels like an analog console, so my learning curve was very short."

Scott Strain
NOISE Studios
San Francisco, California

DA-88 Modular Digital Multitrack

DA-98 Modular Digital Multitrack

MMR Series Digital Recorder and Players

DA-60mkII Time Code Synchronizable DAT

DA-45HR 24-Bit DAT Recorder

OPENPATH

FAXBACK

www.tascam.com

**TASCAM**®

**Take advantage of our experience.**

circle #535 on reader service card

SEE US AT NAMM BOOTH #W1015


the piano will change its relationship to the bass, which may alter the way the bass sounds. So if you make major changes on soloed tracks, be sure to check the sound in the mix immediately.

## KEEP IT UNDER CONTROL

When used on acoustic instruments, compressors work to smooth the dynamics of a performance. Contrary to what many people think, electronic instruments have a good deal of dynamic range. You probably won't have to squeeze anything to death as you would with an electric bass-guitar track, but slight compression of certain sounds can tighten up your mix. Synthesized strings generally sound good compressed at a light ratio (2:1 at -6 dB). If you have a live piano track, you may also want to process it with a little compression (-6 dB threshold, 3:1 to 4:1 ratio), especially if the musician performed with a lot of dynamic feeling.

Compression is often used in an electronic instrument mix to blend several synth outputs together by performing a submix of the desired tracks, busing them to a stereo pair, and patching a compressor across the two channels. Many people use this method to combine sampled sounds, particularly if the samples were derived from a variety of musical styles. I often do it with drum sequences if the samples didn't all come from the same kit. This achieves two things: first, it ensures that the kit will sound cohesive; and if I want, it lets me generate an intentionally overcompressed pumping sound across the drums (a typical hip-hop sound).

Gates are commonly used on synth outputs to quiet or eliminate noise if no mixdown automation is available. Getting rid of extraneous noise is a priority; I have heard some really nasty sounds come out of certain inexpensive sound modules. You don't need to gate the outputs if the modules are active throughout the entire song, only if there are extended periods of inactivity.

Although you could use an expander for this purpose, a gate doesn't affect the dynamics of the performance the way an expander does. Make sure that the gate's attack and release times are set properly; otherwise you may cut off part of the performance.

Finally, for those folks producing urban music, it's not a bad idea to patch a limiter across the stereo bus—just enough to catch peaks from the kick drum. After all, you don't want people getting mad at you because your mix blew their woofers out. (For a more detailed look at using dynamics processors, see "Conquering Peaks" in the December 1998 **EM**.)

## A TOUCH OF 'VERB?

In general, electronic sounds designed to emulate acoustic instruments should receive the same multi-effects treatment as their acoustic counterparts. A piano should be processed like a piano, regardless of where the sound

### THE DADDY OF DANCE

For the most part, dance music is created with electronic instruments; break beats, loops, sequences, and samples often constitute close to 90 percent of a dance track. So who would know more about mixing electronic instruments than a dance-music mix engineer?

Chris Rivera has been involved in the New York dance-music scene for almost ten years. He recently finished mixing a single for the German dance band Electrik Kloud that will be released in the United States this summer. I caught up with Rivera while he was on vacation and picked his brain on mixing.

**How does a dance-music mix differ from, say, a rock mix?**

There are a lot more rules for people doing rock production. When I'm mixing, I don't have to worry about things like making sure the piano sounds full. I can pretty much do whatever I want within reason—of course, the mix still needs to sound good. In general, though, I have a lot more creative liberty with dance music.

**What element do you usually build a dance mix around?**

Definitely the beat. When you're on the dance floor, you need to feel the pumping of the kick. Once your beat is rocking, you can start bringing in supporting tracks, like synth pads. If there are any vocals, they're usually the last tracks I bring in. I know that goes against conventional techniques, but it's how we do it. Vocals just aren't all that important with dance music.

**What factors contribute to poor electronic-instrument mixes?**

Poor placement of instruments within the mix. I've heard a lot of mixes from guys just getting started in this business—their transient samples come in right on top of the synth pads. All of a sudden, the pad disappears, and you're focused on this sample. Then, once the sample has passed, the whole mix sounds empty. You need to have a place for everything in the mix.

**How much multi-effects processing do you use?**

Actually, very little. If I've been involved with a project from the beginning, I try to choose sounds and mold them so that they won't need a lot of effects processing in the mix. Sometimes I'll use an autopanner on a pad or a tap delay in certain places for effect, but generally I keep the mix fairly dry.

**What's the best advice you can impart about mixing dance music?**

Hook up with someone who owns or works at a club. Print a DAT of your mix and bring it down to the club so you can check it out in the most important listening environment. Sure, your mix has to sound good on the radio, too, but you're really mixing for the people at the club. So go down there, crank up the system, and see how it sounds.


No-compromise stereo reverb in combination with as many as 4 additional stereo effects, 200 superbly crafted presets, database sorting, and complete flexibility of routing and effect order in every program make the MPX 1 an indispensable recording and performance tool.

Powered by two separate DSP processors, the MPX 1 is loaded with 56 Pitch, Chorus, EQ, Modulation, Delay and world-class Reverb effects – each with the audio quality and control flexibility you'd expect in a dedicated processor. The 200 presets, designed for a wide variety of performance, sound design and production applications, exploit the unique characteristics of each effect – and a built-in database function makes it easy to find the right program fast.

An interactive front panel gives you instant access to each effect and its essential parameters, as well as push-button control over tempo, morphing, and mix and level settings of any or all effects. Whether you're looking for a rotary cabinet, a 4-Band Parameteric EQ, Ducking Delay, Pitch Shifting, or virtual rooms of any size and description, the MPX 1 is the right tool for the job.

*For studio effects on the road, put an MPX 1 in the effects loop of your stage rig and an MPX R1 on the floor. A single cable provides stomp-box control of MPX 1 effects.*

**Heard In All The Right Places**

lexicon

A Harman International Company

SEE US AT NAMM BOOTH #645

Lexicon Inc., 3 Oak Park, Bedford, MA 01730-1441
Tel: 781/280-0300 Fax: 781/280-0490
Email: info@lexicon.com Web: www.lexicon.com

circle #536 on reader service card


originates. (A little room reverb is a nice touch on a piano, incidentally.)

Now, you can really have some fun with original sounds. Nobody is going to criticize you for processing a Kosmic Kazoo with too much chorus! But watch the spatial placement of the effects. Just because you have the returns panned hard left and right doesn't mean that the effect itself is located on the outer edges of the mix; it might be located somewhere in between, which could cause a conflict with one or more of the other mix elements.

## FROM THE GROUND UP

When approaching a mix, I suggest you start by determining where you want to place everything on the stereo field and then pan the tracks accordingly. Push all the faders up, and get a rough mix going. Next, use EQ to tweak the sounds in context with the rest of the tracks so that nothing is clashing. If you hear something funny with any of the tracks, solo that track, isolate the problem, and fix it. When you're done, pull all the faders back down.

Next, determine which track (or tracks) to mix the song around. Most people agree that you should build a mix around the most important element—whatever will sell the song. Doing this ensures that you won't be caught with a great mix that has absolutely no room for the prized track. So generally, pop music (especially ballads) should be mixed around the vocal; jazz around the soloist; and rock, urban, and alternative around the rhythm instruments (drums, loops, and bass).

Bring the volume fader of your focal-point track up to about 80 percent. (If you decided to mix around the rhythm section, start with the kick drum and bass guitar.) Apply whichever multi-effects processing you want, but don't obsess over the levels of the effects returns; they'll need to be readjusted anyway once you start adding more tracks to the mix. Then start bringing in the rest of the components, adding effects where needed.

If you're building around the rhythm instruments, follow the kick drum and bass with the snare and other drum tracks. Then bring in supporting instruments (piano, guitars, strings, and so on), followed by percussion, lead instruments, and solo instruments. Finish up by adding background vocals, samples, and sound effects.

When building around a solo instrument or vocal, I find it best to bring in some sort of accompaniment first, like acoustic guitar or piano. Follow that with the rhythm instruments, as outlined above, and finish with the supporting cast. If all your levels are good, you should still feel the energy of the first track you put up even after you've added all the other instruments. Don't forget to check the mix in mono, especially if you think your recording could be broadcast; many radio and TV stations do not broadcast in stereo.

Next, address the tracks with dynamics processors where needed, make any necessary EQ changes, and adjust the levels of the effects returns. Finally, automate your tracks and print to tape. It looks easy on paper, doesn't it?

## A LITTLE MASTERING

Once you're reasonably happy with what you hear, you'll want to establish the mix's overall frequency parameters. How much high end do you want? How much low end? True, the mastering engineer usually takes care of these things, but most professional mix engineers will use a parametric or graphic EQ across the stereo bus before printing the mix to 2-track. This allows them to set high and low boundaries for the mix—a particularly smart move when working with electronic instruments, where the frequency responses of the tracks can run the gamut. (It will also help you EQ and set the levels of the extremely high- and low-frequency instruments.)

Again, find a CD with similar content and audio quality to the project you're working on. Listen to the overall volume of the upper and lower frequencies. Then compare that CD with the mix you have going, and make minor adjustments to the stereo EQ where needed. A graphic EQ—I prefer the dbx 2231—is an excellent tool for this application.

A lot of hip-hop, dance, and R&B music has extremely heavy low-end content. This contributes to these genres' distinctive sounds, but loud low end doesn't equal *good* low end. In other words, don't boost 9 dB at 120 Hz during mastering to get the kick drum to stand out more; go back and fix it in the mix. A good mix should require very little tweaking at the stereo bus.

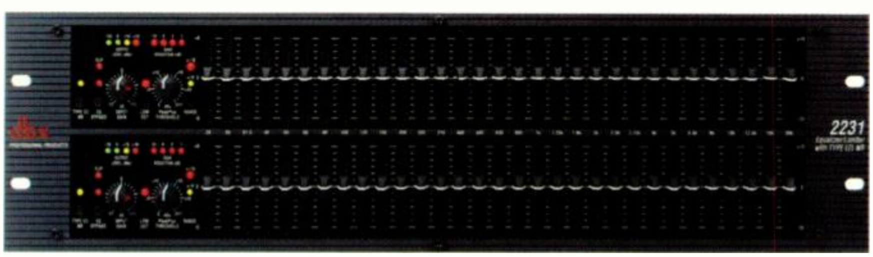
Finally, a little compression (-3 dB, 1.5:1 ratio) across the stereo bus can compensate for subtle level changes that you may not have caught in the mix. Alternatively, as I mentioned earlier, limiting may be in order. (Be sure to set your threshold just below the peaks you want to eliminate.)

## BOOGIE DOWN

The most important thing you can do for any mix is put it to rest once you're done. Let it sit for a few days, allow your head to clear, and then listen to it with fresh ears. At that point, you'll probably want to make a few thousand adjustments, but that's fine. What's essential is that you take a break from the project.

When all is said and done, a mix of electronic instruments employs many of the same techniques as a mix of acoustic instruments. In fact, an electronic mix actually allows you to be *more* creative. If you keep in mind the basic principles I've outlined here, you should be able to construct a solid, three-dimensional mix that jumps right out of the speakers.

**EM** *Associate Editor* **Jeff Casey** *recently turned a hip-hop song into a country tune with a 4-band parametric EQ.*




**Graphic equalizers, such as this dbx 2231, are invaluable mastering tools.**

# The Birth Of A Legend

### The Diaphragm

All GT Electronics AM Series mics use a super-thin 3-micron, gold-evaporated mylar diaphragm for the ultimate in sensitivity and transparent response. Over 1" in diameter, the extra-large diaphragm is precisely tensioned by hand to ensure perfect accuracy.


### The Tube

The GT AM61 and AM62 maintain an all-tube signal path for preamplification. The tube, a military-spec GT5840M, is the most advanced audio vacuum tube ever made. It offers incredibly low self noise, a long life and nearly non-existent microphonics. The result...classic, warm, rich tone that you demand from a legendary tube mic.

What turns a studio microphone into a legend?

Natural **warmth**. Rich character. Crystalline clarity that perfectly captures the fine details of every **nuance**. Most importantly, it provides an indescribable intimacy that makes listening a deeply emotional experience.

Legendary mics are made by people who share a passion for the art of sound. So we gathered together some of the most **passionate** designers in the world to create GT Electronics, a new division of Alesis that's dedicated to the qualities of **legendary** audio equipment design. It all starts with the AM Series large diaphragm studio condenser microphones, which offer **everything** you'd expect from a legendary mic except the legendary price.

**GT**. The new legend has arrived.

*Fixed Cardioid*

AM51
CLASS A FET

*Cardioid, Omni, Figure 8*

AM52
CLASS A FET

*Fixed Cardioid*

AM61
TUBE

*Cardioid, Omni, Figure 8, Super-Cardioid*

AM62
TUBE

**GT Electronics - A Division of Alesis Corporation**



GT Electronics 1633 26th Street Santa Monica CA 90404 www.gtelectronics.com

the new legend

**SEE US AT NAMM BOOTH #1033**

circle #537 on reader service card

Improve the sound of your personal studio in three easy steps.

# The Well-Tempered Studio

BY GEOFFREY GOACHER

Good acoustics are often the dividing line between professional and personal studios. After all, the gear that's used to record and mix much of what we hear on the radio these days is quite similar to what can be found in a well-appointed personal studio. Largely, it's those big, beautiful, acoustically accurate rooms for recording and mixing that separate the big guys from the little. But by giving a little attention to the acoustics in your own studio, you can improve the quality of your mixes so that they compete with the best the majors have to offer.

ILLUSTRATION BY ROBERT PIZZO

1.


crowded house


## BASIC CONSIDERATIONS

Before getting into acoustics, make sure that there are no weak links in your monitoring system. Monitoring systems for critical listening must have a fairly flat frequency response from about 60 Hz (or lower) to 16 kHz (or higher). The power amp should also have as flat a frequency response and as low a distortion spec as possible. Fortunately, most of the studio-grade near-field monitors and power amps on the market today meet these specifications. Therefore, selecting the "right" system is often just a matter of personal taste.

The monitoring system must be set up symmetrically within the room. The distance between the speakers should be the same as the distance from each speaker to your ears, thus forming an equilateral triangle with your head. For near-field monitoring, your speakers should be about two to four feet apart, depending on their size and dispersion and what is most feasible ergonomically. Also, the center of this equilateral triangle should be equidistant from the room's side walls (see **Fig. 1**).

Unfortunately, falling short of sonic accuracy is common, even when high-quality gear is placed symmetrically in the room. The overall sound is often boomy and muddy, the bass is too loud or too soft, the high-end is dull or harsh sounding, and the imaging is blurry and undefined. Room acoustics can play a significant role in creating, and reducing, these problems.

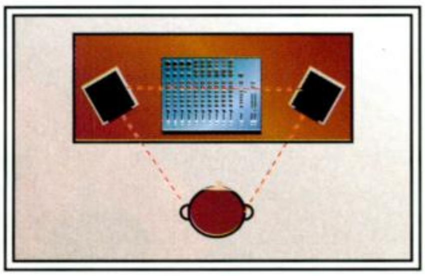


**FIG.1: Studio speakers should be placed symmetrically within the room, forming an equilateral triangle with your head.**

In pro studios, room acoustics are considered a top priority. Typically, owners spend lots of money on professional consultation, premium construction, and first-rate sonic treatments, sparing no expense to achieve problem-free, acoustically "neutral" monitoring environments. However, overcoming acoustical problems is not outside the financial realm of the personal studio owner. You should expect to spend at least the same amount of money for acoustical treatment as you did for your monitors.

The acoustical problems that occur most commonly in small monitoring rooms are room resonances (standing waves), speaker/boundary interference, early reflections, and poorly diffused late reflections. These problems can be overcome in three easy steps.

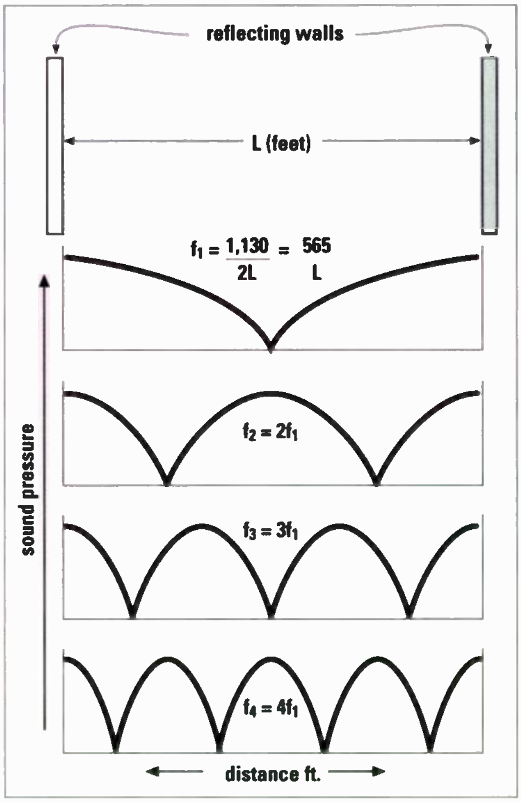



**FIG. 2: When wavelengths are twice as long as the distance between the room boundaries, a "standing wave" reinforcement of the wavelength occurs at that frequency and its harmonics, causing a boost in their volume.**

# Step 1:

## CONTROLLING RESONANCE AND REFLECTIONS

The first step deals with low frequencies—from 20 Hz to 500 Hz. This frequency range affects the smoothness of the bass and low mids: if the room's acoustics are balanced, the bass and low mids will be full and warm; if the room has significant frequency boosts in this range, the sound will be boomy or muddy; and if the room has significant frequency dips in in this range, the sound will be thin and hollow. The goal in Step 1 is to flatten out the room's low-frequency response so as to avoid erroneously mixing music to compensate for the boosts or dips caused by the acoustic environment.

**Resonance and standing waves.** The way low frequencies behave in a room is dictated largely by the room's dimensions. Certain frequencies, due to the lengths of their respective sound waves, are reinforced as they move between the room's boundaries (walls, floor, and ceiling), creating resonant boosts in volume at those frequencies (see **Fig. 2**). These resonances are commonly referred to as *standing waves*.

You can estimate the most prominent resonant frequencies of a room by using the following equation:

$$f_1 = 1{,}130/2L = 565/L$$

In this formula, $f_1$ represents the resonant frequency, and *1,130* represents the speed of sound in air under "normal" conditions, which are defined as one atmosphere of pressure at sea level at 21 degrees Celsius. *L* represents the length of the room in feet. For example, if the room is ten feet long, there will be a natural resonant volume boost in the room at 56.5 Hz. In addition, natural boosts in volume will occur at multiples of this frequency: $f_2$ = 113 Hz, $f_3$ = 169.5 Hz, $f_4$ = 226 Hz, and so on.

These resonances become more closely spaced and their volumes diminish as you move up the frequency spectrum. Therefore, in small rooms, resonances are typically not as problematic above 200 Hz.

**Speaker/boundary interference.** Because low frequencies are omnidirectional by nature, they reflect from all nearby room boundaries. These reflections adversely affect low-frequency

For manufacturer information, please see Contact Sheet, p. 227

# Perfect Pitch. Now in 7 delicious flavors.

"...a Holy Grail of recording... Bottom line, Auto-Tune is amazing."
NICHOLAS BATZDORF, RECORDING MAGAZINE

new version
Now supports Pro Tools 24 Mix with full automation
Pro Tools NT version coming soon!


"Here is one product that you absolutely have to have... I have never encountered a device that is as remarkable as the Auto-Tune plug-in... It's completely amazing!"
PETE LEONI, PRO-REC MAGAZINE ONLINE

"The ATR-1 saves us an unbelievable amount of time. It goes with me wherever I go."
AL SCHMITT, PRODUCER

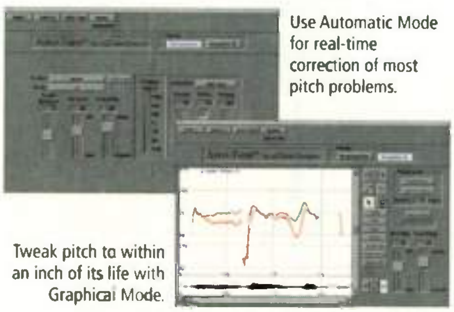
"...the best pitch-correction program is Antares's Auto-Tune ...I use that program to death..."
GEORGE DUKE
MUSICIAN/COMPOSER/PRODUCER

REMEMBER THE NOT-SO-GOOD OLD DAYS? Like way back in early '97, when capturing that perfect vocal — the one with emotional power and rock-steady pitch — still often meant hours of frustrating retakes. Or the time-consuming process of comping a track from many separate takes. Or both.

Well, in the two short years since its introduction, Antares's Auto-Tune™ pitch-correction software has, quite literally, revolutionized the practice of recording vocals. Using Antares's proprietary DSP technology, Auto-Tune corrects intonation problems in vocals or solo instruments, in real time, without distortion or artifacts, while preserving all of the expressive nuance of the original performance. With audio quality so pristine that the only difference between what goes in and what comes out is the intonation.

## Celebrating Diversity

Back when Auto-Tune was available only as a Pro Tools™ plug-in, we knew of quite a few people who bought kilo-buck Pro Tools systems just to be able to run Auto-Tune. Today (as a quick scan of our bashful spokesmodels will show), there are versions of Auto-Tune compatible with virtually all the leading computer-based digital audio platforms.



Use Automatic Mode for real-time correction of most pitch problems.

Tweak pitch to within an inch of its life with Graphical Mode.

And for those who don't work in a computer environment — or just want to free up their computers for other tasks — we offer the ATR-1 Auto-Tune rack processor.


## The Professional Standard

Auto-Tune and ATR-1s are used daily by thousands of audio professionals around the world. Whether to save studio and editing time, ease the frustration of endless retakes, or save that otherwise once-in-a-lifetime performance, Antares pitch-correction is the overwhelming tool of choice. Whatever your favorite flavor, check it out at your Antares dealer or call us for a free demo CDROM. And prepare to be amazed.

ANTARES
REALLY COOL STUFF FOR MAKING MUSIC

SEE US AT NAMM BOOTH #1715

ANTARES AUDIO TECHNOLOGIES 464 Monterey Avenue, 2nd floor, Los Gatos, CA 95030 | www.antarestech.com
US and Canada: 888 332 2636 | from overseas: 408 399 0008 | info@antarestech.com

circle #538 on reader service card

# The Well-Tempered Studio

response, making the bass sound as though it's coming from different directions (see **Fig. 3**).

These slightly delayed reflections of the original signal cause comb-filtering peaks and dips within the range of frequencies above the modal resonance range (typically 200 Hz in a small room) to an upper limit of approximately 500 Hz. The increasingly directional nature of sound above 400 Hz makes speaker/boundary interference less of a problem for mid and high frequencies.

Standing waves and speaker/boundary interference can cause frequency-response deviations as high as 15 dB. This amount of level variation could keep you guessing about proper levels for all the bass frequencies you mix.

**Identifying the problems.** The best way to tell if you have a problem with standing waves or speaker/boundary interference is through a combination of listening and measurement. First, listen to a finely engineered CD through your monitoring system at a decent mix level. The CD you select for this exercise should have a tight bass sound and minimal reverb. Some of my favorites are Tchad Blake's mixes on Crowded House's *Woodface* album and Sheryl Crow's two latest albums, *Sheryl Crow* and the *Globe Sessions*. Tchad Blake uses interesting imaging and minimal reverb, which makes his mixes great for critical listening exercises. (But then, that's the style of music I mix; you may prefer something else).

As you listen to these CDs through your system, notice whether the bass sounds tight, smooth, and consistent in volume. If the mix sounds full and warm, then your room naturally promotes good bass response. Larger "small rooms," rooms with lots of windows, and rooms with lightweight walls tend to balance bass frequencies nicely.

However, if the room rumbles and booms with the music, or if the bass either sounds mushy or alternates between high and low levels, then you may have a problem with resonance or speaker/boundary interference. At this point, it's a good idea to take a precise measurement of your room's resonant characteristics.

**Simple measures.** Measuring for room resonance and speaker/boundary interference requires a high-resolution frequency analyzer, rather than the usual octave or ⅓-octave real-time analyzer. Octave and ⅓-octave analyzers average out too much information to be useful for this task. Fortunately, some new software programs allow you to perform high-resolution acoustical analysis affordably from a computer (see the sidebar "Acoustical Programs for the PC").

For example, **Figure 4** depicts a high-resolution, low-frequency response graph made with AcoustiSoft's *ETF* 4.0 software. Measurements such as these can help you better identify the problem areas in your studio. Given the high resolution of the measurement, narrow notches in the response aren't that bad, but you should pay attention to the general frequency-response trends. Notice, for instance, that the average signal level is around -16 dB, with a boost of 8 dB centered around 300 Hz, a 12 dB boost around 125 Hz, and one sharp 16 dB boost around 55 Hz.

Based on this measurement, you can guess that our monitoring system would sound muddy in this room because of the 300 Hz boost, too bassy because of

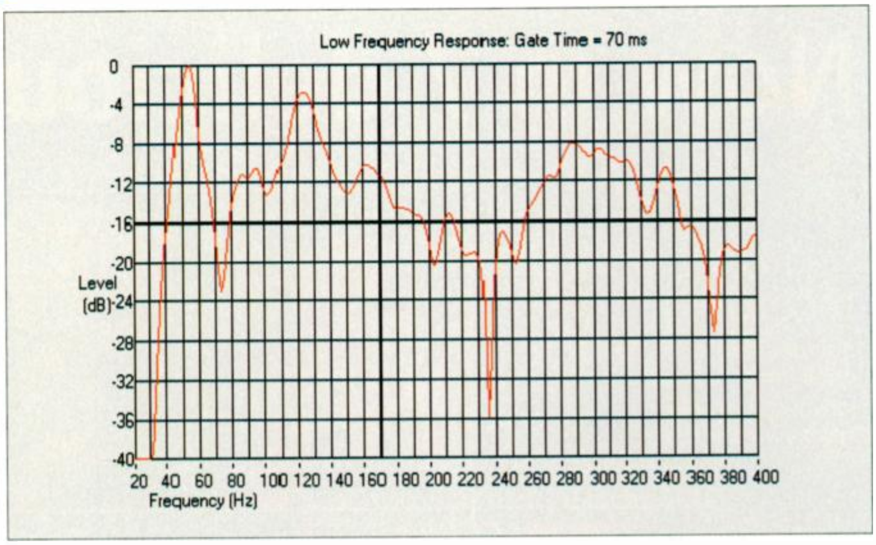



**FIG. 4: Shown is a high-resolution, low-frequency response graph made with AcoustiSoft's *ETF Room Acoustics Analyzer* 4.0 software.**

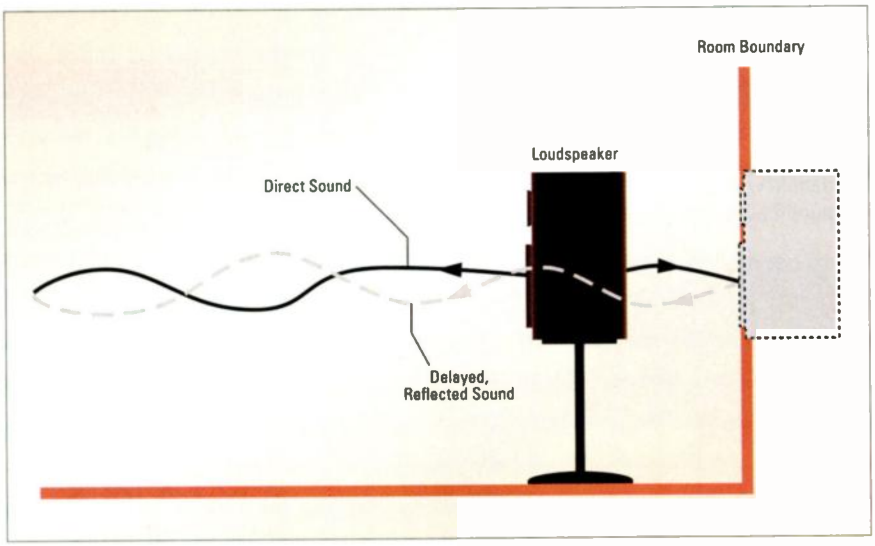
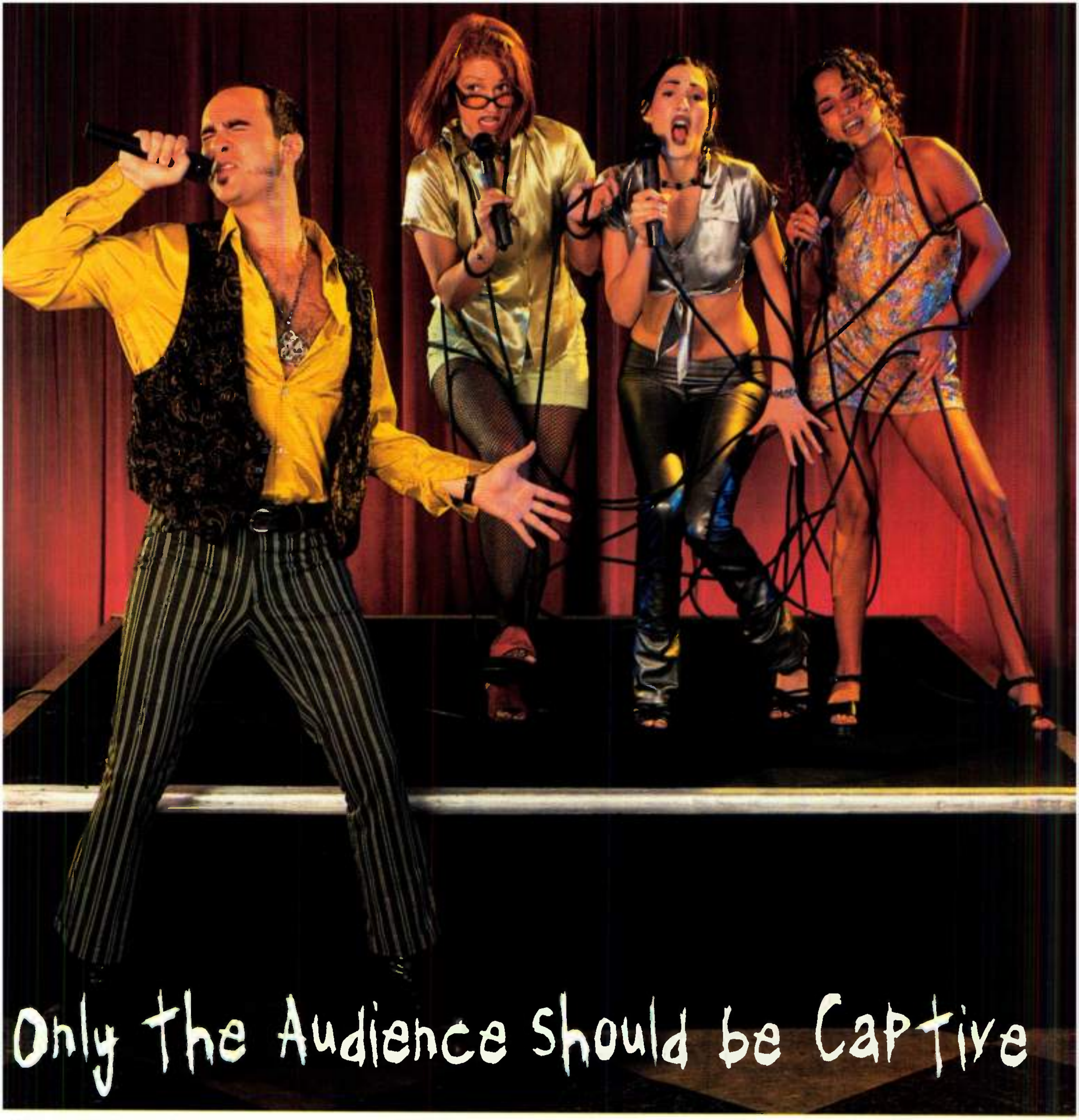



**FIG. 3: The omnidirectional nature of low frequencies causes them to reflect from all nearby room boundaries.**




If you think wireless technology is out of reach for the rest of your group, it's time to rethink your options... Jensen Wireless Systems deliver megaband looks, feel, and sound at garage band prices. It's Jensen's Emancipation Proclamation: freedom from wires for all! From the club circuit to weekend jam sessions, Jensen brings professional-quality sound and dependability together with wireless technology and affordable prices to rock your world.

- Wireless systems for every need or budget
- Wireless mics, guitar transmitters and in-ear monitor systems
- Transmit up to 16 channels simultaneously
- Up to 40 frequencies to select from
- Street prices from the low- to mid-$200 range
- Guaranteed Jensen quality

Let There Be Freedom

JENSEN
Pro Audio
WIRELESS

Jensen Music Industries, A division of RECOTON AUDIO CORPORATION
Call toll free (877) 863-5548 www.jensenmusic.com

the 125 Hz boost, and too boomy because of the peak at 55 Hz. Given that the specifications for the loudspeakers used in this test are flat throughout their low end, it can be assumed that these low-frequency response deviations result from room influences.

**Fixing the problems.** A good way to smooth out resonance and boundary-reflection problems is by optimizing the location of the speakers and the listener in the room. Resonance and boundary reflection are less pronounced in certain areas of a given room. In fact, changing the location of the loudspeakers and listening position often results in a drastic change in sound from the previous location.

Optimizing speaker and listener locations used to be a process of trial and error. Now, however, there are several PC-based programs that can model the acoustics of your room and help you find a location where room resonance and boundary reflections are minimized.

I used a speaker/listener optimization program on the same monitoring system shown in **Figure 4**. Then, after changing the location of the listening station, I remeasured the low-frequency response (see **Fig. 5**). As you can see, the optimization program worked well: the average signal level is still about -16 dB, but the significant boosts shown in **Figure 4** have been smoothed out. The only exception is that the boost at 55 Hz remains.

The frequency response in **Figure 5** still appears to have a lot of variation because of the high resolution of the measurement. If measured using a lower resolution of ⅙-octave or ⅓-octave, however, the frequency response would appear as a nearly flat line.

After optimizing the placement of the speakers and listener, you can do several other things to further reduce any problematic boosts in the low frequencies. Applying normal acoustical foam works well to dampen high- and mid-frequency energy but doesn't adequately absorb low frequencies. You can absorb low frequencies by using a *bass trap*, which is any acoustical device that absorbs low-frequency energy in a room.

Often, the best place to put bass traps is in the corners of rooms, because that's where low-frequency energy collects. As the low-frequency energy is absorbed, the various peaks (as exemplified in **Figures 4** and **5**) are reduced, resulting in a smoother bass sound overall. The average listening room benefits from having about 1 percent of its total volume dedicated to bass trapping.

Many companies manufacture broadband bass traps, but one in particular, Acoustic Sciences Corporation (ASC), also offers affordable acoustical consulting for treatment of low frequencies using a test that they developed called the M.A.T.T. (Musical Articulation Test Tones) test.

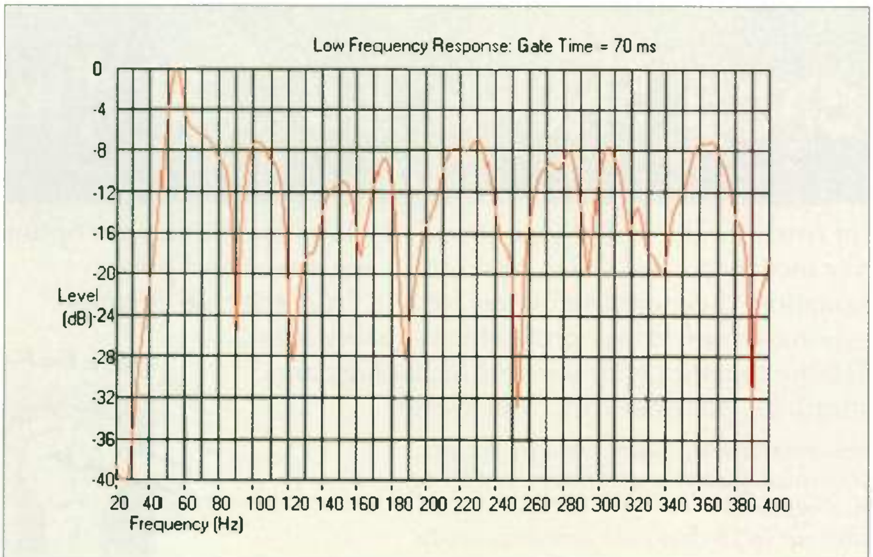



**FIG. 5: This is a low-frequency response graph of the same room after speaker/listener placement was optimized.**

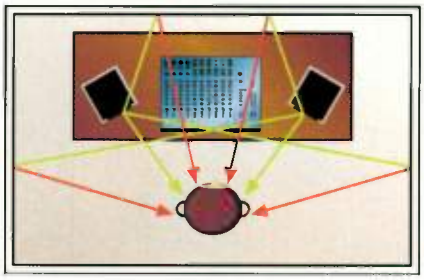


**FIG. 6: The direct sound (green) and any early reflections (red) are heard as one sound.**

# Step 2:

## REDUCING EARLY REFLECTIONS

The second step deals with frequencies of 500 Hz and up. This range has a critical effect on the accuracy of the monitoring system's imaging and its mid- and high-frequency tonality. The biggest detriment to mid- and high-frequency accuracy is the presence of early reflections.

**Early reflections.** When listening to your monitors, you hear a combination of the direct sound from the speakers followed by the reflections of the direct sound from the room's boundaries (walls, ceiling, and other hard surfaces). Reflections that hit the ear within 20 milliseconds of when the direct sound is produced are heard as part of the direct sound and are called *early reflections*. Because sound waves travel at a rate of about one foot per millisecond, most of the first reflections that make their way to the listening position in a small room qualify as early reflections (see **Fig. 6**).

Early reflections often add audible comb-filter distortion to the direct signal, tainting the frequency response with a variety of boosts and dips. Early reflections also tend to blur the stereo imaging between the speakers, making it difficult to accurately hear the exact position of sounds within the stereo field.

**Identifying early-reflection problems.** The best way to determine whether you have a problem with early reflections is to listen for and measure them. For this exercise, play a well-mixed CD that has clear and precise imaging, such as one of those mentioned in Step 1.


As you listen, notice whether the locations of the instruments are clearly identifiable in the stereo spread or whether they blend between the speakers. You should be able to hear the various instruments coming from specific points in the stereo field. Problematic early reflections, however, will degrade the aural clues that help us identify stereo imaging, and the resulting mix will sound blended and fuzzy.

Measuring for early reflections requires a high-resolution analyzer that can generate an impulse response, or an energy-time curve, of your environment. Or you can use one of several acoustics analyzer programs mentioned in the sidebar "Acoustical Programs for the PC" to generate an energy-time curve. This kind of measurement will give you a clear idea about any problems you might have with early-reflection levels.

In **Figure 7**, the direct sound from the speakers is shown at 10 milliseconds, with the early reflections occurring between 10 to 30 milliseconds—a period of 20 milliseconds. In a balanced acoustical environment, the reflections between 10 and 30 milliseconds would be 15 to 20 dB below the level of the direct sound. In other words, early reflections should be virtually inaudible.

The early reflections are about 15 to 18 dB below the level of the direct sound for nearly 15 milliseconds. Although not perfect, this particular room would have good imaging and would be free of significant comb-filtering in the mid- and high-frequencies. Prior to this test, the room graphed in **Figure 7** had already been treated with acoustical foam at strategic points to reduce early reflections. If measurements or listening tests confirm that your room has problematic early reflections, you should consider a similar treatment with acoustical foam.

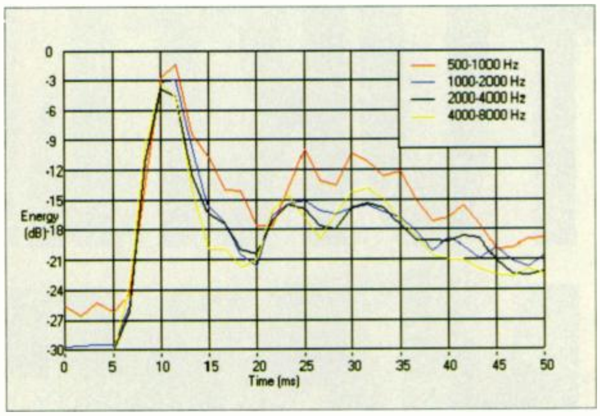



**FIG. 7: An energy-time curve. The direct sound is shown at 10 milliseconds. The reflections in this graph, from 10 to 30 milliseconds, are the early reflections (a total period of 20 milliseconds).**

**Fixing early reflection problems.** The goal in fixing early reflections is to reduce them to an inaudible level, which is typically about 15 to 20 dB below the level of the direct sound. This is where sound absorption materials, such as acoustical foam, work wonderfully. Companies such as ASC, RPG, and Acoustical Solutions market a variety of sound absorption products. Generally, it is best to use products that have sound-absorption coefficients greater than 1 and that absorb frequencies down to 400 or 500 Hz.

You don't need to cover every inch of your walls with this stuff: using too much absorption can make the room sound too dry. Rather, determine the best places to put sound absorbers to reduce early reflections. You can easily do this by using the "mirror trick."

To perform the mirror trick, sit in the mix position facing the speakers. Then, have someone move a picture-size mirror flush along the walls, ceiling, and other surfaces to your sides and front. (You are allowed to turn your head, of course.) Any spot on the walls or ceiling where you can see the face of the speaker in the mirror should be covered with sound-absorption material. If you are unable to see the front of the speaker in the mirror, it is best to leave the surface alone. Once you get the concept, you can perform this operation without a mirror. It's also a good idea to cover the wall space behind and between the speakers with sound absorption to reduce any diffraction reflections from the speakers.

There will be a noticeable improvement in your system's imaging and frequency response once you have reduced early reflections in your room. You will immediately hear sounds in your mixes that were previously masked.

## ACOUSTICAL PROGRAMS FOR THE PC

High-quality acoustical measurement systems have traditionally been too expensive for the average audio enthusiast. These systems have therefore remained in the domain of acoustical consultants and audio designers, who have bigger budgets. Recently, however, significant advances have been made in the availability of affordable measurement systems for home-studio owners.

AcoustiSoft's *ETF Room Acoustics Analyzer*, Liberty Instruments' *LAUD*, and JBL's *SMAART* are all affordable, Windows-based software programs that will turn your computer into an acoustics analyzer. Each of these companies can give you further instruction on taking and interpreting measurements of your room if you need help.

Several new programs are also available for optimizing location of monitors and the mix position. KB Acoustics' *Visual Ears*, Pilchner-Schoustal's *Acoustics-X*, and RPG's *Room Optimizer* are three PC-based programs that will model the resonances and boundary reflections of your room and help you find an optimal location for loudspeakers and listeners. These programs usually work very well, provided your room fits the program's criteria.

# Step 3:

## DIFFUSING LATER REFLECTIONS

The third step, which also covers frequencies of 500 Hz and up, deals with *late reflections*. Unlike early reflections, late reflections arrive outside of the ear's integration time and do not


necessarily affect the accuracy of the monitoring system. In fact, late reflections are desirable for creating acoustical "spaciousness" in the room. Without them, the room would sound like a dry, anechoic chamber.

The problem is that small rooms have such a low density of reflections that later reflections typically sound sparse and choppy in their decay. You can improve this situation by increasing the diffusion in the room.

The concept of diffusing late reflections in small rooms proceeds from the fact that mid- and high-frequency sounds typically reflect from a flat surface at a single angle only. However, when mid- and high-frequency sounds strike a diffusing surface on a wall (such as a quadratic residue diffuser), they reflect back into the room at many different angles (see **Fig. 8**). This results in a more complex spread of sound, which is known as *diffusion*. Spreading the reflections out in space and time also reduces their volume levels.

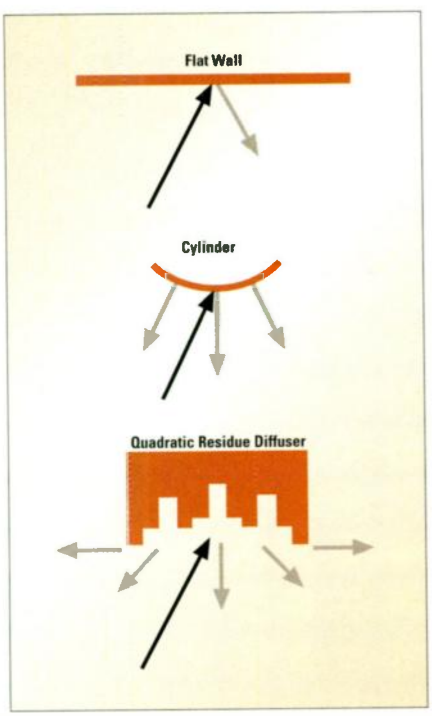



**FIG. 8: Diffusers and cylinders scatter reflections in many different directions.**

The best way to identify diffusion problems is by listening for them. Sit in the mix position, and have another person clap loudly in front of each speaker. This simulates the sound of transients coming from the speakers. Does the room have a noticeable echo at the mix position? Do the reflections sound well blended, or do they sound harsh and fluttery? If you notice echoes, your room would benefit from added diffusion.

**Fixing diffusion problems.** Improving diffusion is generally done by placing diffusive surfaces along the back wall of the room. For example, when sound strikes a cylindrical (as opposed to flat) object, it reflects into the room laterally over a 120-degree arc. This creates a uniform spreading of the reflection back into the room. Diffusers can be as simple as bookshelves or cylindrical objects, or as complex as primitive root and quadratic residue diffusers. However, some diffusers are much more effective than others.

The most effective diffuser is the quadratic residue diffuser. First conceived and proposed by acoustics researcher Manfred Schroeder, it was commercially introduced into the audio world by Dr. Peter D'Antonio of RPG Diffusor Systems, Inc. A quadratic residue diffuser is essentially a box that comprises a series of parallel "wells" of varying depths. The depth and width of the wells are calculated to give an effective diffusion of a specific range of frequencies. In addition, these units reflect sound laterally over a 180-degree angle.

The primitive root diffuser is also highly effective. Its well configuration is based on a different mathematical sequence than that of the quadratic residue diffuser. Both these types of diffusers are commercially available through RPG Diffusor Systems.

Placing a diffuser at each point where sound first reflects from the back wall will improve diffusion and will result in a more natural and "spacious" decay. Plan on covering about 60 percent of the rear wall with diffusers if you want to achieve a highly noticeable diffusion effect in the room.

Diffusion is like icing on the cake for room acoustics: it gives the room a pleasant, spacious ambience that often makes the room much easier to work in. Once you've completed this last step, your room's configuration will most likely resemble the one shown in **Figure 9**.

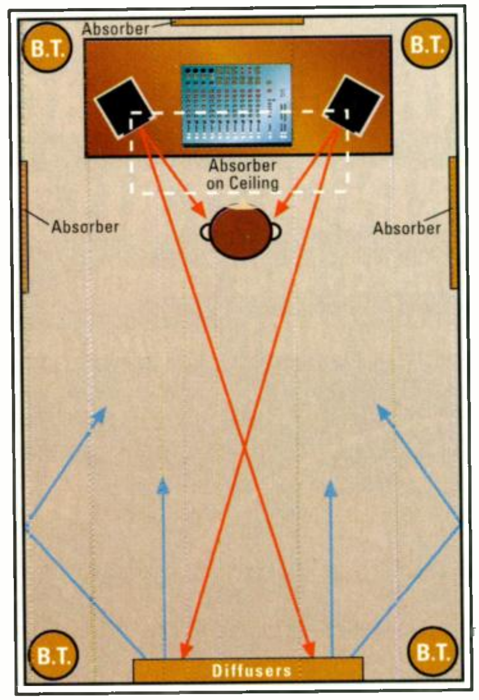


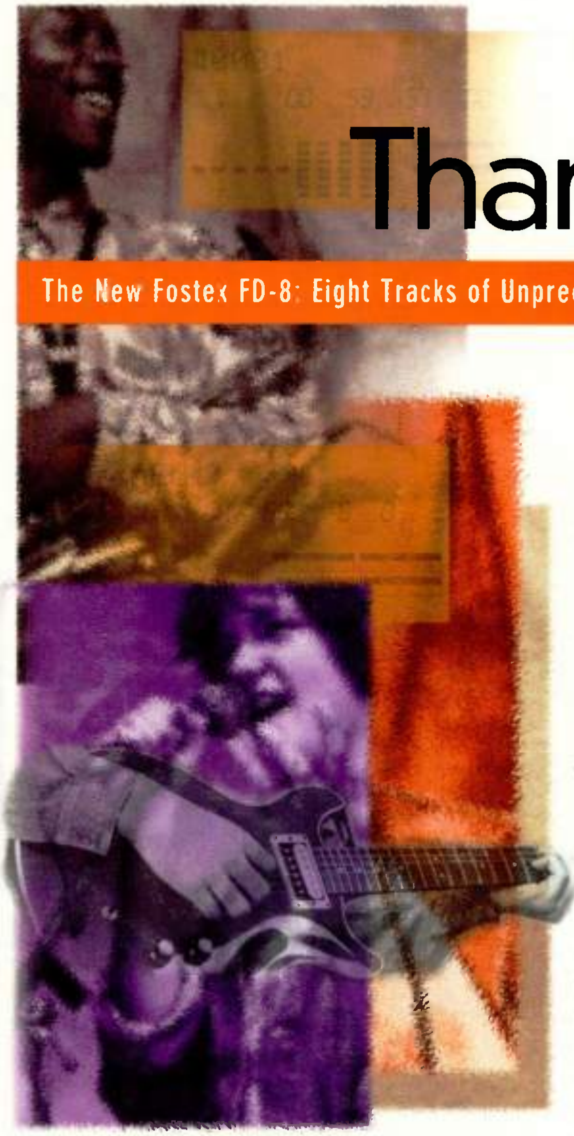

**FIG. 9: The use of corner bass traps, strategically placed absorbers, and back-wall diffusers can significantly improve room acoustics and monitoring capabilities.**

## UPON REFLECTION

An accurate monitoring system in a balanced acoustical environment allows you to clearly hear the imaging, tonality, and other nuances in your mixes. Smoothing out room resonance and boundary reflections, reducing early reflections, and diffusing late reflections are the best methods of improving a studio's listening environment. As you go through each of these steps, take advantage of the various tools mentioned in this article, and don't be shy about contacting people for advice.

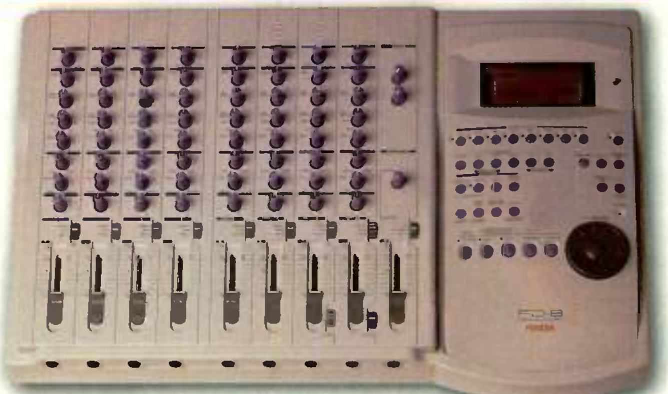
In the end, good acoustics will not only make your music and your mixes more fun to work with and listen to, they will also increase your efficiency and make the task of mixing easier. Ultimately, you will turn out more reliable and professional-sounding mixes—and more of them.


**Geoffrey Goacher** *is the founder of Acoustical Research Associates, which specializes in research and communications on audio and acoustics for critical-listening environments.*



# Way More Than Four More

**The New Fostex FD-8: Eight Tracks of Unprecedented Flexibility and Power—Priced at Under $900**



You could say that the new Fostex FD-8 is a duplicate version of Fostex's groundbreaking FD-4 digital recorder plus 4 channels more. But you'd be overshadowing the FD-8's amazing list of powerful, professional features. Because in addition to offering 8 channels of pure Fostex digital audio, the FD-8 also gives you two independent recording modes (one maximizes sound quality, the other recording time), 16 Virtual Tracks (for a total of 24 recording tracks), "Digital Track" inputs and outputs, and much more. Even full MIDI implementation. Plus, just like the FD-4, it gives you the freedom to choose your own optional SCSI recording media from a wide variety of available removable and hard drives. And with an M.S.R.P. of only $899, it shines a whole new light on what you can expect from a portable digital multitracker.

**ADAT LIGHTPIPE™**
I/O jacks provide best solution to exchange up to eight tracks of data with an ADAT™.

**Random Access EDITING**
Full Cut/Copy/Paste/Move plus Undo and Redo.

**44.1kHz Sampling**
Uncompressed 44.1kHz, 16-bit linear, CD-quality digital audio.

**DUAL XLR Mic Inputs**
With Mic Trim on channels 7 and 8 for use with low-Z microphones.

**S/PDIF Optical I/O**
Input digital signals from other sources and output directly.

**105dB Dynamic Range**
Analog 8 IN x 2 OUT familiar mixer with 105dB dynamic range.

**3 BAND EQ Plus Sweep**
3-point EQ (2 shelving & 1 parametric types) on all 8 channels.

FD-8 LIMITLESS
Fostex

Don't need a full eight channels? The original limitless multitracker, the FD-4, gives you four channels of pure, "no-compression" digital recording and many of the professional features you'll find in the FD-8. It's the only four-track multitracker that lets you choose your own recording media from a variety of removable and hard drives. And—best of all—you can get your hands on one for under $600!

Fostex Corp. of America • 15431 Blackburn Ave. • Norwalk, CA 90650
Tel: 562-921-1112 • Fax: 562-802-1964 • http://www.fostex.com • e-mail: info@fostex.com

Fostex

SEE US AT NAMM BOOTH #1155

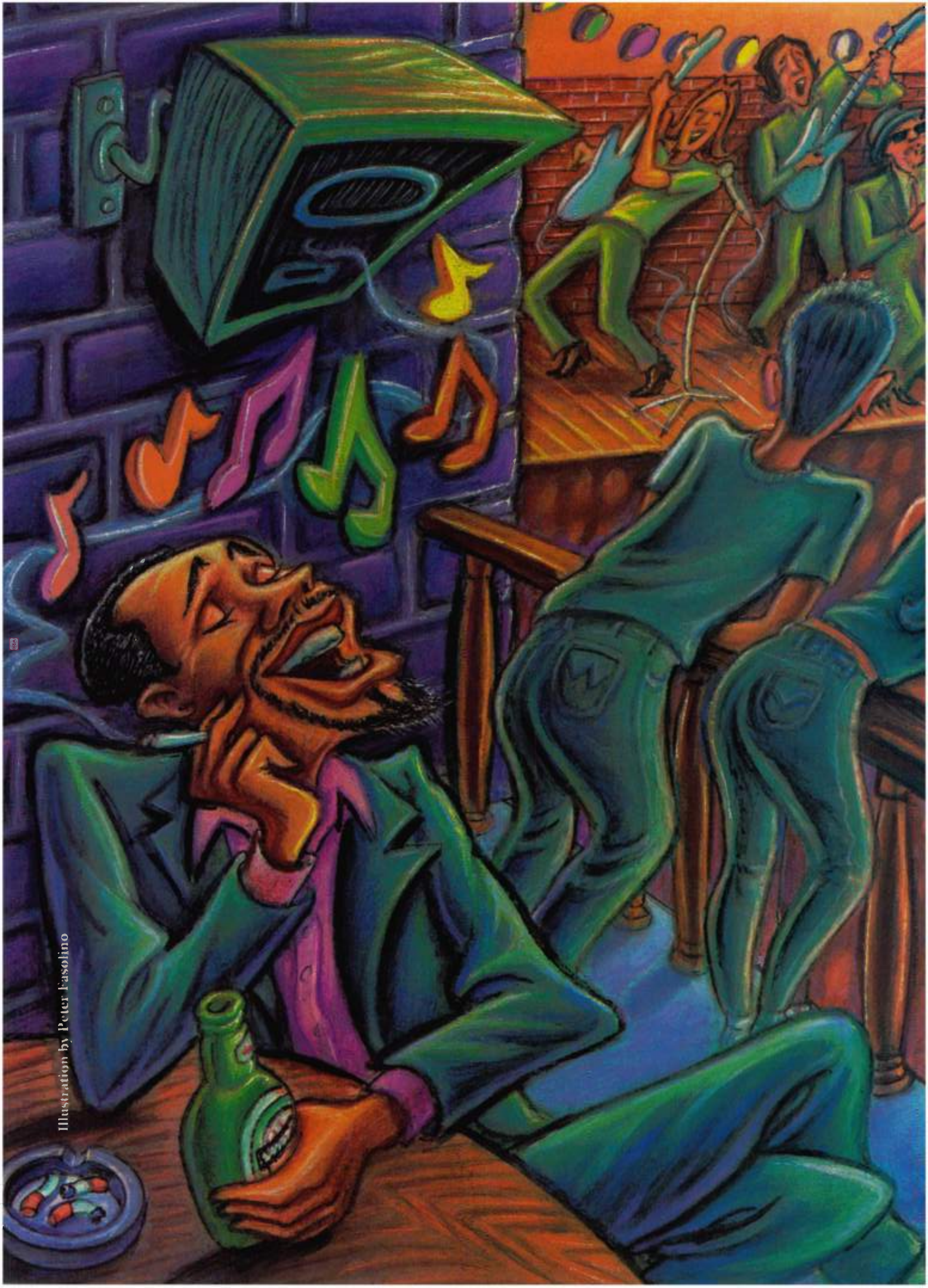


Illustration by Peter Fasolino

# Playing to the Back of the House

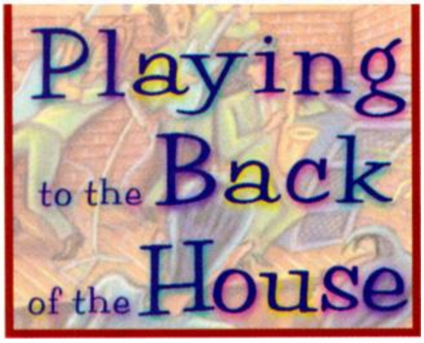
By Rudy Trubitt

Ideally, a live sound sytem would deliver the same good sound to every member of the audience, regardless of the listener's location in the room. The unfortunate reality is that never happens. Even with great equipment, a cooperative band, and an engineer's good judgment, it's rarely possible to deliver comparable sound to audience members positioned both near to and far from the stage.

What's a band to do? In some cases, you can use additional speakers to improve sound for those parts of the audience that are poorly covered by your main-stage P.A. cabinets. Properly deployed, using a distributed loudspeaker system can mean delivering better sound to your entire audience. But don't imagine that you can simply daisy-chain a few extra speakers from the same amplifier that's running your main cabinets. Without separate amps and a high-quality digital delay line, extra speakers spread around the room are sure to make your music sound worse rather than better.

A distributed speaker system helps you deliver quality sound to every seat in the house.

Fortunately, with the right equipment and technique, you can improve your sound in acoustically poor or larger rooms. But before we begin, a note of caution: try these ideas only when you have extra setup and sound-check time before your performance—you'll need it.




## What's the Problem?

Live sound is plagued by two gremlins. The first is the acoustical properties of the performance space. Most venues are acoustically challenging, to say the least. Excessive reflections, low ceilings, and awkward floor plans all cause their share of grief.

The second problem is that loudspeakers (woofers and tweeters) do not radiate all frequencies in all directions equally. As the frequency produced by a particular speaker driver gets lower, the radiating pattern of the projected sound widens. The flip side of this phenomenon is that as frequencies get higher, they are directed forward in an increasingly narrow pattern, or "beam." As a result, the perceived sound of the cabinet changes depending on whether you are standing in front of it or off to the side.

This combination of difficult room acoustics and typical speaker-system characteristics can wreak havoc. Reflective, reverberant rooms rob clarity and definition from the sound-reinforcement system, and loudspeaker cabinets compound the problem by projecting clear sound straight ahead but uneven sound off to their sides. This means that much of the sound reflected off the walls, floors, and ceilings is not only delayed but also colored the instant it leaves the cabinet, due to the aforementioned frequency-response anomalies in the cabinet's off-axis response.

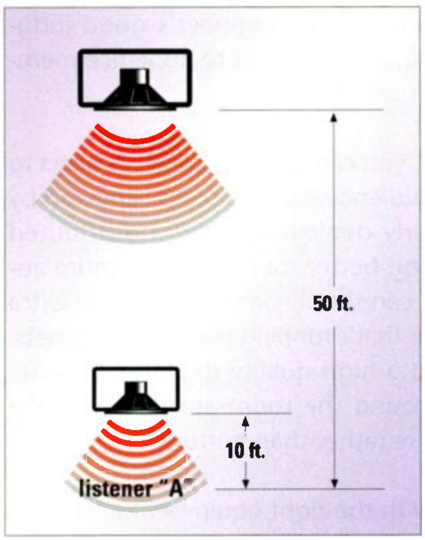



**FIG. 1: When the same sound is played by two speakers separated by a distance, the listener (A) will first hear the sound of the nearer cabinet. The delay time before the sound from the distant cabinet reaches the listener is determined largely by the distance between cabinets (40 feet).**

The easiest way to address both issues is to seat each member of the audience relatively close to, and directly in front of, a loudspeaker cabinet. Sitting close to the speaker improves the ratio of direct to reflected sound, and being on-axis ensures optimum frequency response. Of course, this is much easier said than done.

A quick way to visualize your sound system's coverage in a room is to walk through the audience area while periodically glancing at your horn tweeters. Ideally, you'll be able to look down the throat of a horn from nearly every vantage point. In the real world, this won't always be the case. Depending on the size or shape of the venue, it may simply be impossible to cover every seat with your stage-left and stage-right main speakers.

In these instances, consider adding supplemental speakers to project additional sound to areas of the house that would otherwise be "dark." Although accomplishing this requires careful planning, it can be better than cranking up the sound so that it reaches the entire room, but blows out the people up front, angers the management, and still delivers loud, muddy sounds in the back of the room.

## Delayed Speaker Concepts

Before putting up additional speakers to improve audience coverage, you have to consider issues raised by a simple fact of physics: the speed of sound. Ignoring this critical point and just sticking up extra speakers around the room guarantees worse sound than your main speakers would have provided by themselves, so pay close attention.

Sound travels more slowly than you might expect, which causes serious problems for sound-reinforcement systems. At sea level and with nominal humidity conditions, the speed of sound is 1,130 feet per second. As a quick and dirty rule of thumb, just remember "one foot per millisecond."

The speed of sound becomes critical when any member of your audience is listening to two or more speakers that are playing the same program material. This is because the sound from one speaker will probably arrive earlier than the sound from the second speaker. This delay between arrival times causes serious perceptual problems, and the greater the *difference* in the two speakers' distance from the listener, the larger the problems become. Put another way, if you are standing in the coverage pattern of two speakers, one near and the other farther away, the sound is likely to be poor.

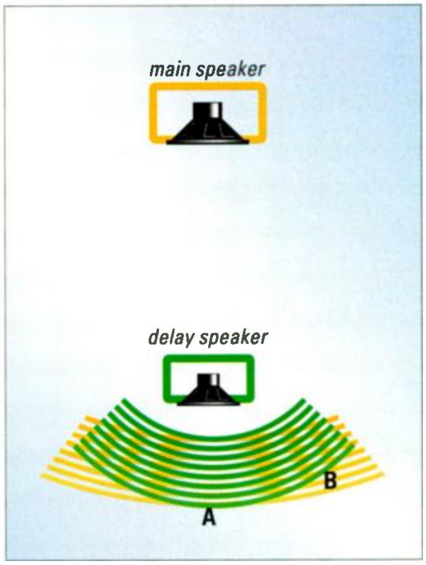



**FIG. 2: By introducing a delay equivalent to the distance between speakers, the sound from both cabinets can arrive at listener A's location at the same instant. Here, color is used to identify the source of each set of sound waves: the yellow waves come from the main speaker and the green from the delay speaker. The interference pattern occurring between the yellow and green sound waves causes comb filtering for listener B.**

Consider the illustration shown in **Figure 1**. Here, the listener (A) is 50 feet from the stage and 10 feet from an extra speaker positioned to help the people in the back hear. Thus, the stage speaker is 40 feet farther from the listener than the extra speaker is.

The problem is that electricity travels faster than the speed of sound. The sound waves from the stage speaker take about 50 milliseconds to arrive at listener A's location, and by the time they arrive, the "same" sound waves from the rear speaker have already come and gone. The difference in distance (40 feet) translates into a delay slightly over


## Gold Channel

**DIGITALLY ENHANCED MIC PRE-AMP**

The TC Electronic Gold Channel is a Digitally Enhanced Microphone pre-amplifier and a DSP signal refinement toolbox. Plug in your microphone, connect the Gold Channel's outputs to any analogue or digital recorder, and safely capture your signal in the best possible recording quality.

## M2000

**STUDIO EFFECTS PROCESSOR**

The Engineering Group at TC Electronic was given carte blanche to create the optimal studio effects processor. Being musicians and studio engineers themselves, they have a feel for what is needed in modern high-grade processors. The library of effects includes: Reverb, Pitch Shift, Delay, Chorus, Ambience, Equalization, De-essing, Phasing, Compression, Gates, Expansion, Limiting, and Stereo Enhancement.

## M3000

**STUDIO REVERB PROCESSOR**

Setting the new industry standard with the VSS3 technology the M3000 is the best sounding, most versatile and easiest to use professional reverb today and well into the future. Combining the ultimate control of directivity in the early reflections with a transparent and harmonically magnificent tale, the art of reverberation is brought to at new and higher level.

## FIREWORX

**MULTI EFFECTS PROCESSOR**

Want to breathe some fire into your music? Get ready for FireworX! An explosive Multi Effects Processor. The FireworX' broad palette of effects includes: Vocoder - Ring Modulation - Synth Generator - Formant Filter - Resonance Filter - Multitap Delay - Reverse Delay - Reverb - Dynamics - Chorus - Flange - Phase - Pitch Shift - Fractal Noise and more ...

## Finalizer EXPRESS

**STUDIO MASTERING PROCESSOR**

The Finalizer Express is the fast and efficient way to turn your mix into a professional master! Based upon the TC Electronic Multi-Award winning Finalizer Mastering Technology, it delivers the finishing touches of clarity, warmth and punch to your mixes, putting the world of professional mastering within your reach.

## Finalizer PLUS

**STUDIO MASTERING PROCESSOR**

The Finalizer Plus gives you the extensive and complete range of controls you need to add the finishing touches to your mix. Compared to the Finalizer Express the Finalizer Plus offers an even wider range controls allowing you to fine-tune every aspect of the mastering process.

**t.c. electronic®**
ULTIMATE SOUND MACHINES

**circle #542 on reader service card**

TC ELECTRONIC INC., 790-H HAMPSHIRE ROAD, WESTLAKE VILLAGE, CA 91361, USA · PHONE: (805) 373 1828 · FAX: (805) 379 2648
FOR LITERATURE CALL TOLL-FREE: 800 288 5838 E-MAIL: INFOUS@TCELECTRONIC.COM · HTTP://WWW.TCELECTRONIC.COM/FINALIZER
TC ELECTRONIC A/S, SINDALSVEJ 34, DK-8240 RISSKOV, DENMARK · PHONE: + 45 8621 7599 · FAX:+ 45 8621 7598

# Sweetwater will pay you TOP DOLLAR to trade in your old recorder towards a brand new 20-bit Alesis ADAT Type II Multitrack!

You know you want it! The incredible, affordable new ADAT Type II recorders just got a whole lot more affordable thanks to this exclusive upgrade offer available only from Sweetwater Sound!

That's right, we'll buy any ADAT, tape or hard disk or analog recorder and pay you top dollar in trade towards a new Alesis 20-Bit ADAT Type II system!

Have you been wanting the benefits of 20-bit recording but didn't think you could swing the investment? With our exclusive upgrade program, we make it extremely economical to sell us your existing recorder and get the 20-bit Alesis system you want—without the hassle of placing want ads, putting up signs, etc. What could be easier?

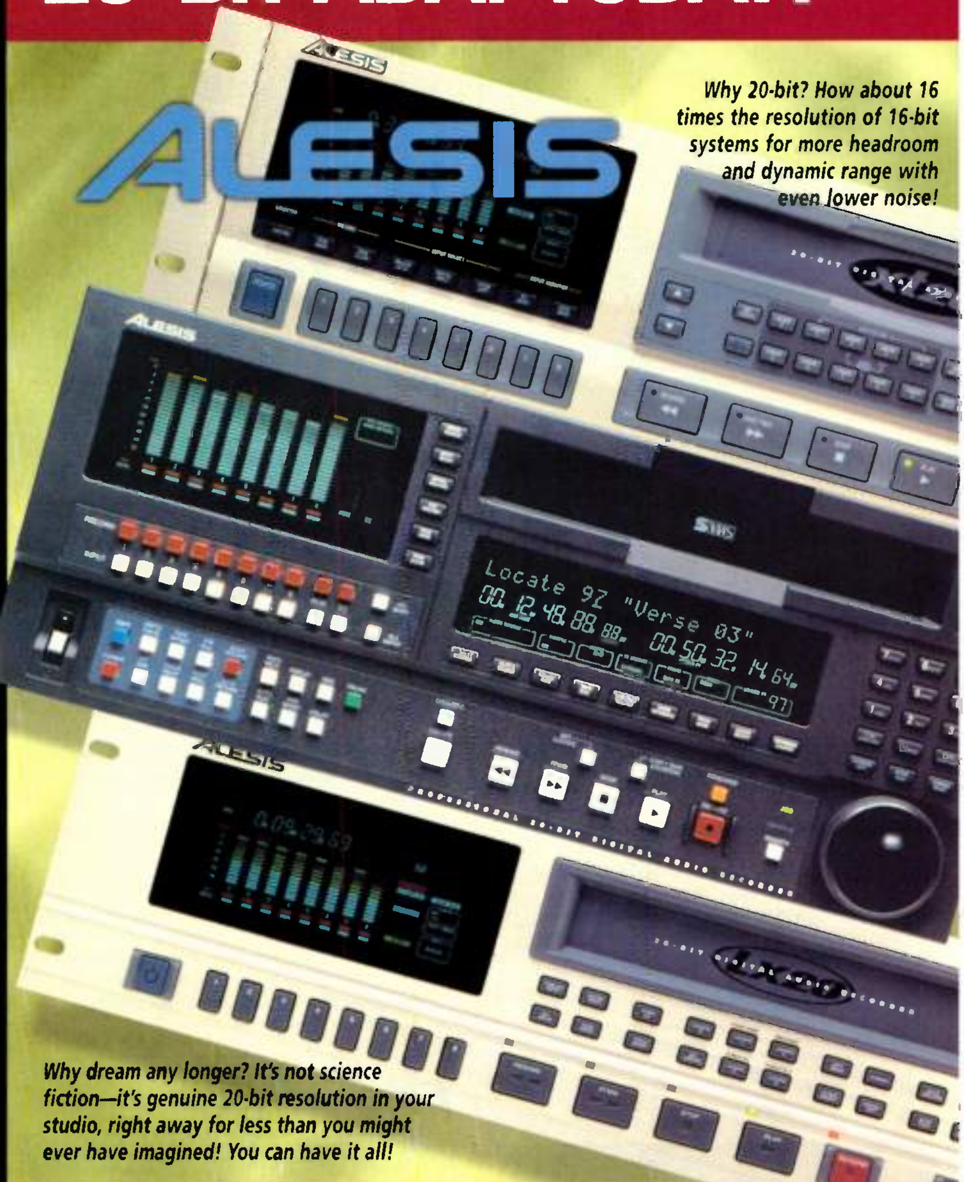
And there are 3 great 20-bit ADAT Type II recorders to choose from! Want the ultimate in low-cost, pro-quality recording? Choose the LX20. The "work-horse" of the Alesis 20-bit line is the powerful XT20. And for top installations where only the best will do, consider the amazing M20. No matter which 20-bit ADAT you choose, you get the phenomenal sound quality of true 20-bit recording. Of course, your new 20-bit ADAT Type II will play any current 16-bit ADAT tapes.

Why wait another minute to enjoy 20-bit ALESIS ADAT TYPE II quality? Take advantage of this unique offer from Sweetwater Sound today!

## *NOBODY KNOWS ALESIS LIKE SWEETWATER!**

*Want some proof? What other retailer has a full page ad in this magazine offering to pay you money when you upgrade to 20-Bit!

# WHY WAIT? YOU CAN UPGRADE TO AN ALESIS 20-BIT ADAT TODAY!




*Why 20-bit? How about 16 times the resolution of 16-bit systems for more headroom and dynamic range with even lower noise!*

*Why dream any longer? It's not science fiction—it's genuine 20-bit resolution in your studio, right away for less than you might ever have imagined! You can have it all!*

## *MUSIC TECHNOLOGY DIRECT — and the Best value Guaranteed!*

**CALL (800) 222-4700 RIGHT NOW AND UPGRADE TO 20-BIT ADAT QUALITY!**

(219) 432-8176 • FAX (219) 432-1758 • sales@sweetwater.com
5335 BASS ROAD • FORT WAYNE, INDIANA 46808

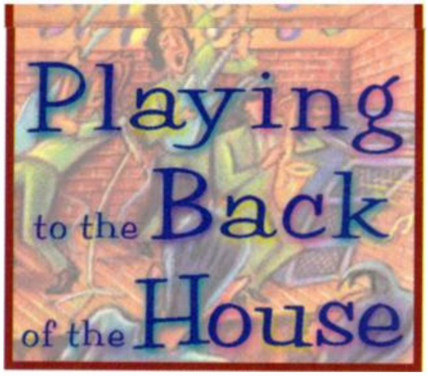
Win **FREE GEAR!** Go to "www.sweetwater.com/contests" right now for more info!

Sweetwater SOUND INC.
www.sweetwater.com

SEE US AT NAMM BOOTH #1827

circle #543 on reader service card


A PERFECT PAIR
TC Electronic Gold Channel and Audio-Technica AT4060
What do you get when you cross the most amazing new mic preamp/dynamics processor/EQ with a pair of the best sounding, lowest noise new tube microphones?
You get more than exceptional sound. You get more than extraordinary flexibility — from pristine digital to warm valve flavor.
You get the most talked about, lusted after and in demand pair of studio tools of the year.
Sure, you could use a $500,000 console and a pair of $20,000 vintage mics. Or, for about a hundredth of the investment, you can have a pair of AT4060s and a Gold Channel in your studio tomorrow morning!
AUDIO-TECHNICA AT4060
• Tube, Large Diaphragm Condenser
• Low self-noise
• High SPL
audio-technica®
t.c. electronic
ULTIMATE SOUND MACHINES
TC ELECTRONIC GOLD CHANNEL
• Two Channel Mic Preamp
• Full Dynamics Processing and EQ
• 24 Bit Conversion / Up to 96kHz Internal Processing
MUSIC TECHNOLOGY DIRECT — and the Best value Guaranteed!
No matter what gear you already own, The Gold Channel and AT4060s will take your studio and your music to stunning new heights. Why not call us right now and start enjoying the tremendous power and awesome sound of this perfect pair immediately?
Call (800) 222-4700
(219) 432-8176 • FAX (219) 432-1758 • sales@sweetwater.com
5335 BASS ROAD • FORT WAYNE, INDIANA 46808
Sweetwater SOUND INC.
www.sweetwater.com
circle #645 on reader service card
SEE US AT NAMM BOOTH #1827




location. Again, there is no ideal solution for this problem. Redefining the delays to be more correct for any single audience position simply makes the delays less accurate for another location nearby.

Optimizing the delay times for a straight line measurement between main and delay speakers is often the best compromise unless most of the listeners are sitting off-center from this imaginary center line. In that case, you must choose a nominal center point in the audience area covered by the delay speaker and measure the distance from that point to each speaker (main and delay). The difference between these two measurements should be used to calculate the delay time.

## Bad Ideas

Although it is impossible to make the delays perfect for each member of the audience, it is easy to make the problem dramatically worse. Consider the following examples of what *not* to do.

In **Figure 4** we see a delay speaker positioned so that it fires across the back half of the audience area. Although you might be tempted to do this in a small room—placing the rear speaker along a wall and facing it out across the audience—don't do it. While that is appropriate for a multichannel surround-sound system, it will be a disaster for a delayed speaker.

Here's why: consider the position in which listener A is standing. This audience member is very close to the delay speaker, so a substantial delay in the rear speaker would be needed for the main P.A. sound to arrive in sync at the listening position. On the other hand, look at listener B's position. Here, the distance from delay speaker and from the main speaker is roughly equal. In this case, little or no delay would be required to synchronize the two sources. This exacerbates the inherent geometry errors in our previous examples, making this configuration much worse than having no delay speaker at all.

Taking this bad idea to its extreme, imagine placing the delayed speakers at the rear wall of the room, behind the audience and facing the stage. In this case, sound from both front and rear speakers would arrive simultaneously in just one row of the audience seating area. The rest of the audience (and the band) would experience the equivalent of an extra loud slapback from the rear of the room, generated by the rear speakers.

The one time you might want to place delay speakers sideways across a room or at a rear wall is if the audience area is completely out of the coverage pattern of the main speakers, such as in a room with a dogleg (see **Fig. 5**).

Direction of fire is not the only consideration, of course. When positioning delay speakers, don't place them too close to the main speakers because this increases the severity of off-axis listener errors. Also, don't place different delay cabinets at different distances from the stage unless you have a separate delay line for each distance.

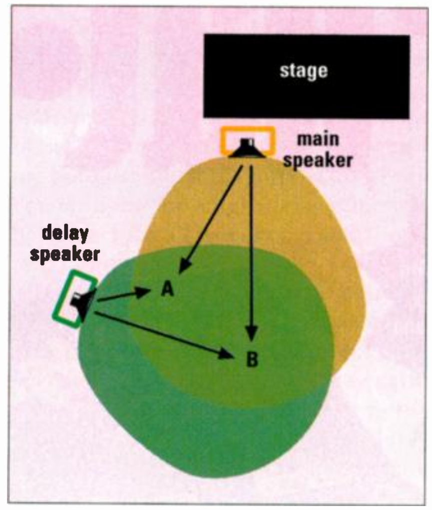



**FIG. 4: Although this speaker arrangement might make sense for "surround" effects, it's a disaster for a delayed speaker system.**

Finally, it's important to avoid aiming main or delay speakers in the direction of nearby large, flat surfaces, such as cinder-block walls or large windows. The sound reflected from these surfaces will cause significant comb filtering throughout the affected audience area.

## Setting Delay Times

Once you've positioned your delay speakers, you'll need to dial in your delay settings. (Obviously, you'll need to get signals from your mixer to the delay speaker system, but that's a separate issue that we'll address shortly.)

Using a tape measure, note the distance in feet from each delay cabinet to its nearest main-stage loudspeaker. Then apply our formula of 1,130 feet per second divided by the number of feet. It's better if the delay speakers are a little late rather than a little early: this way, sound from the stage speakers arrives first, helping the listener focus on the action onstage rather than on the sound from the delay speaker. The level of the delay system is also important in this regard.

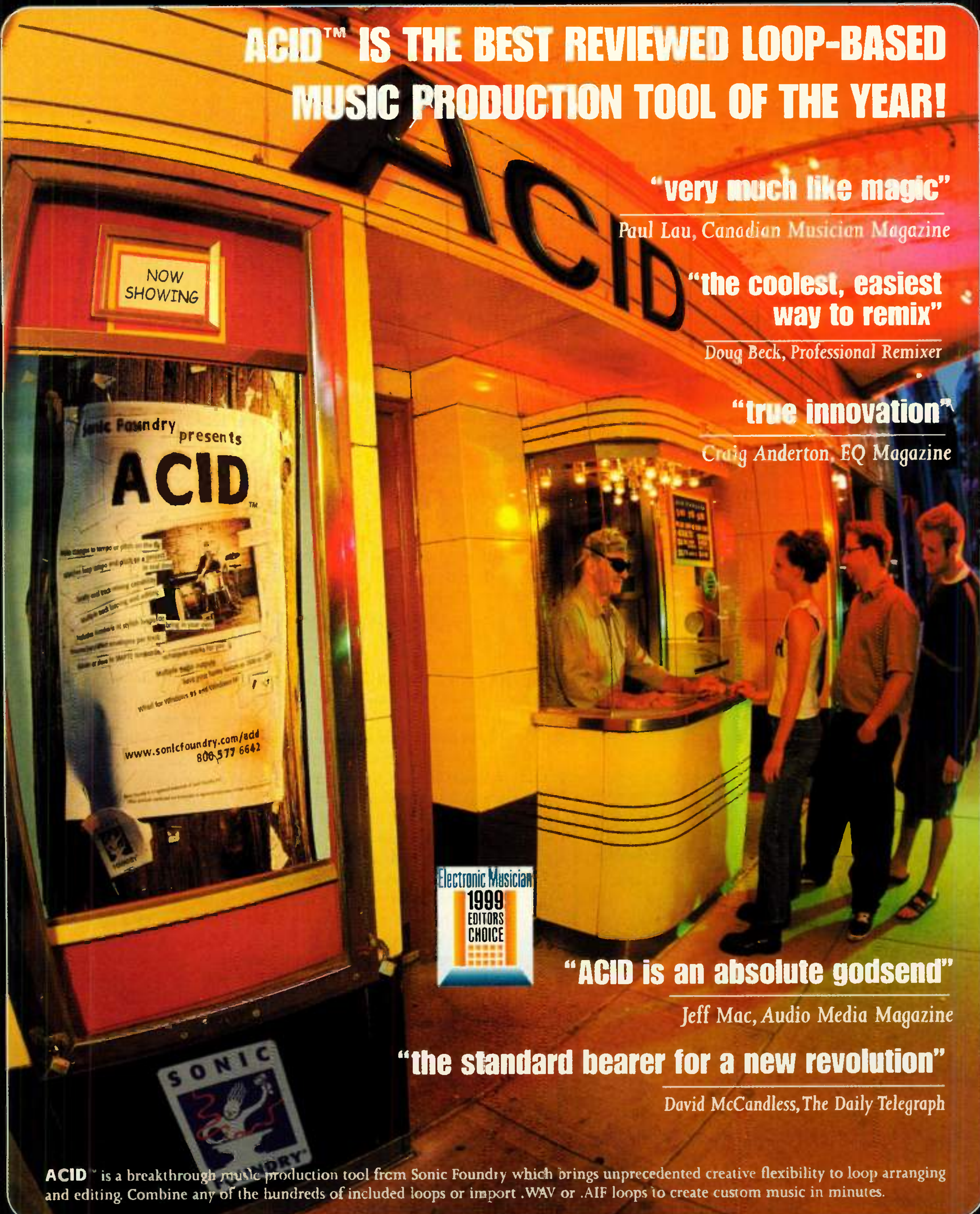
Fortunately, our simplified calculation of one foot per millisecond errs in the right direction, causing the delay speaker to lag behind the sound coming from the mains. However, for distances greater than 40 feet, the error in our easy-to-memorize conversion factor grows large enough that the delay speaker will be too far behind the mains. In such cases, I suggest breaking out the calculator and doing the math.

### FEAR OF FLYING

Elevating your main or delay speakers evens out the distribution of sound and can help ensure unobstructed transmission of sound to each listener's ears. However, remember that raising speakers at or above the height of a person's head can be dangerous: heavy cabinets can and do tip over, endangering life and limb of musicians, crew, and audience alike.

Never suspend speakers overhead without the assistance of a qualified rigging technician. Just because your loudspeaker cabinet has integrated "flying hardware" doesn't mean that you have the skills to hang it safely, nor does it ensure that the building's ceiling can support the added weight.

The simplest way to raise smaller speakers above ear level is to use properly deployed stands designed for cabinets of the size and weight being lifted.




**NOW AVAILABLE EVERYWHERE**

Call 1 800 57 SONIC for a dealer near you
or visit our Web site at www.sonicfoundry.com

Madison, WI 53703 Tel: (608) 256 3133, Fax: (608) 256 7300, CompuServe: 74774,1340 or GO SONIC, Internet: sales@sonicfoundry.com. Sonic Foundry and Sound Forge are registered trademarks of Sonic Foundry, Inc. Other products mentioned are trademarks or registered trademarks of their respective manufacturers.

circle #545 on reader service card

**SEE US AT NAMM BOOTH #W1301**

# Playing to the Back of the House

## Setting Delay Levels

Once your delay times are properly set, you can set the levels for your delay speakers. I believe the most effective technique here is to try to make the delay speakers "disappear." In other words, turn them up only enough so that you miss them when they are turned off. This way, a casual listener might not even notice that they're on. This keeps the listener's focus on the stage, where the performance is taking place. Running the delay speakers too loudly pulls the audience's attention away from the stage, which can be distracting and fatiguing.

An exception to this rule would be delay speakers covering acoustically isolated parts of the venue, such as a room with a deep dogleg or a separate outdoor patio (see **Fig. 5**). If the audience can't see the stage anyway, there's no point in trying to trick them into thinking they are hearing your main speakers, as opposed to the delayed cabinets. In the case of an outside area or separate room, if the stage sound and P.A. are attenuated enough, these "satellite speakers" could run without a delay at all because there may be little, if any, sound from the main system present in these remote areas.

## The Rest of the Gear

As you've gathered, fielding a delayed speaker system will require at least one digital delay. You can use a high-quality multi-effects unit (16 bit, 44.1 kHz or better), but it's vital that the unit offer at least single-millisecond delay resolution. Be sure to set the wet/dry mix at 100 percent wet, with no regeneration (delay feedback).

For professional situations, there are numerous products designed just for speaker-delay applications. These units offer delay times calibrated in feet and inches (or centimeters and meters) in addition to milliseconds, which takes the math out of the setup process. Units like these are more readily found at pro-audio dealers (especially those who serve the sound-contractor installation market), rather than in music stores. Don't expect them to include reverb, chorus, or other effects; these are specialized delays.

In addition to the delay line, you'll need separate amplifiers to run the delay speakers. You may need an extra graphic EQ for the delay system, as well.

To get signal to the delay speaker system from your mixer, you could use a Y cord from your main mixer output to feed the delay system. This means that the same mix is sent to your mains and the delays. However, you might not want the same mix in both places. For instance, a vocal-only mix for the delays might be desirable. There are several ways to pull this off.

You could dedicate an aux send to feed the delayed speakers. This would make it possible to dial various channels into the delay mix. Using a postfader send would mean the delay speakers would follow any mix moves performed on the main faders, which is probably preferable for musical applications.

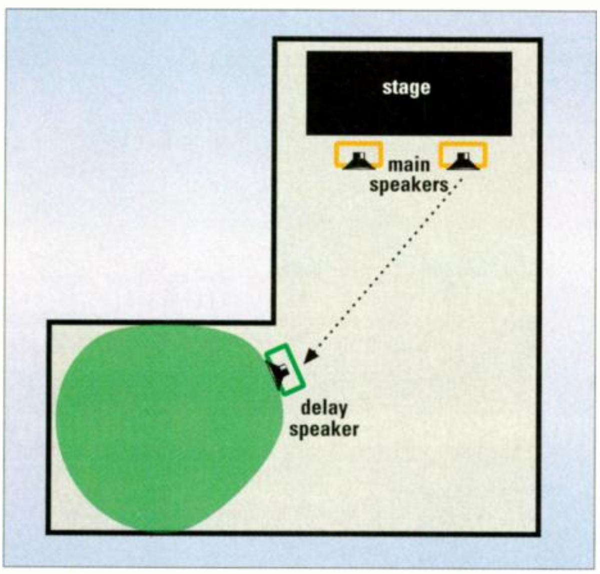



**FIG. 5: In the case of a room with a deep dogleg, a delayed speaker can provide coverage for audience members completely outside the direct coverage area of the main speakers. Choosing this delay time will require some trial and error, since the reflected sound coming around the corner will be delayed relative to the dotted-line direct measurement shown here.**

If you have spare subgroup buses on your console, you could use one of them to feed the delay. Again, these would be postfader outputs, so the delay mix would follow mix moves on the main faders, and the subgroup master would serve as a master delay system level. (Just be sure not to assign this subgroup to the main L/R mix.)

The most flexible method of feeding delays is to use a live-sound console with a mix matrix. This allows each subgroup to be sent in varying amounts to multiple delay "zones." This sort of sophistication is required in large concert systems, in which multiple speaker zones are present, such as a center vocal fill for the first ten rows in addition to a delay zone at the rear of the hall. This degree of complexity, however, is beyond the needs (and perhaps the resources) of most **EM** readers.

## Choosing Delay Cabinets

Ideally, delay cabinets should match your main cabinets tonally; this will help their sound blend smoothly with that of the main speakers. Delay speakers don't, however, need to have the same low-frequency response as the main speakers, because there is likely to be plenty of low end bouncing around the venue from the main system. This means that you don't have to drop large cabinets or subwoofers at the delay speaker positions, unless the delay system is in a separate room, acoustically isolated from the main P.A.

## Final Reflection

Getting good sound is always a challenge. But you can avoid many difficulties by understanding how sound propagates around a room. Once you have a handle on the speed of sound and the interrelated wavelengths of various frequencies, you will begin to see how good sound from your speakers can quickly turn into bad sound in the room. The careful use of delayed speaker systems can be a valuable tool toward the laudable, if unattainable, goal of "perfect sound for every seat in the house."

---

**Rudy Trubitt** *is the author of* Live Sound for Musicians, *published by Hal Leonard. For more information on live sound and other audio-production topics, visit him at www.well.com/user/trubitt. (Thanks to Sabine, Inc.)*

# Steal the Scene with SMFs

***Commercially available MIDI files are opening doors for musicians.***

By Tran Whitley

Professionally created MIDI files of popular music have been around almost as long as MIDI itself. Over the years, these commercial arrangements have been put to good use in a wide range of applications, from educational programs to multimedia soundtracks (see **Fig. 1**). As a longtime producer of commercial MIDI files, I've found that my clientele largely consists of professional musicians who often use MIDI files in highly imaginative ways.

I talk daily with musicians who play in clubs, bars, and restaurants, at parties and weddings, and at other venues. These working pros are always looking for opportunities to improve their sound, enhance their performances, and expand their repertoire. In many cases, Standard MIDI Files (SMFs) provide just the boost they need. As you'll see, the right MIDI files can get you a new gig or even help you keep the gig that you already have.

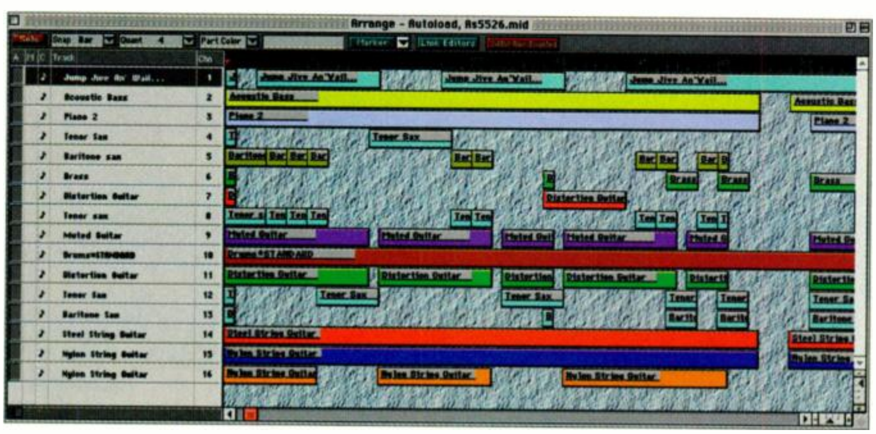



**FIG. 1: Many commercially produced Standard MIDI Files, such as this example from Tran Tracks, involve complex musical arrangements that use all 16 MIDI channels.**

You may be wondering how to get started using SMFs onstage. The short answer is to get a laptop computer, a sequencing program, a good General MIDI (GM) sound module, and a collection of MIDI files, and you're ready to go. However, it's not always quite that simple. You may have to address a few issues in order to tailor your setup to meet your performance demands. For example, do your songs need to play back-to-back? Will they always be in the same order, or do you need random access to the titles? If you are working with a drummer, will he or she need a click track? What kinds of edits (key and tempo changes, solo sections, different endings, alternate arrangements, etc.) will you need?

These considerations could affect the type of hardware and software that you choose and will certainly affect your initial preparations. If your needs are not overly complex, you may be able to replace the laptop computer with a dedicated Standard MIDI File player, such as the Yamaha MDF3 (see **Fig. 2**).

Keep in mind that you'll still need a sequencer to prepare your files for playback with a straight MIDI file player (even if it's just to delete the melody).

# THE GOOD STUFF.



Professionals rely on industry-standard tools from Whirlwind to protect each performance and recording from the frustrating failures and poor sonic results of second-rate components. Built to provide consistent superb results through years of tough use, these products are what sound specialists buy when they can buy what they want — *the good stuff.*


The Hot Box™ is Whirlwind's top-of-the-line DI for sending your pure guitar sound direct to the mixer. The ultra-high input impedance, super-wide bandwidth and vanishingly low noise mean you won't find better sonic performance anywhere.


The Director® is a premium quality DI box with an input select for instrument or amp/preamp, a ground lift switch, and a filter to roll off amp noise.

The IMP® 2 brings a high-quality DI to musicians on a budget. Rugged, good sounding, with a ground-lift switch.


The Mic Eliminator® is a specialty DI that takes your direct guitar, amp, or preamp signal, and adds the clean sound of a classic SM57 + Jensen 12" mic + speaker setup for that famous "miked" sound without the mic.


The Selector® switches one instrument to two inputs or two instruments to one input using a patented totally silent Whirlwind switching circuit.

The MD-1™ is a battery powered studio-quality mic preamp with a transformer isolated line driver output and a built-in headphone monitor. The heavy steel box, 18V phantom power and transformer isolation make it a favorite with sports broadcasters.


The MicPower® is a convenient portable phantom power supply that operates from two standard 9V batteries to power condenser mics.


High Performance PM Tubes® are manufactured for Whirlwind using the latest techniques and specialty metals. If you own a tube amp or effects device, listen to the difference these tubes will make.


The Qbox® is an all-in-one battery powered audio line tester. It includes a built-in mic, speaker, tone generator, headphone amp, and voltage detect LEDs for verifying phantom or intercom power.


The Tester is the industry's favorite continuity tester, detecting shorts, opens, or cross-wiring on cables with virtually any combination of XLR, 1/4", and RCA phono connectors.

**whirlwind**™

99 Ling Rd. • Rochester • NY • 94549 • TOLL FREE 888.733.4396 • FAX 716.865.8930 • http://www.WhirlwindUSA.com

circle #547 on reader service card

Also, some keyboard workstations (such as the Roland XP-80 or Kawai K5000) provide real-time SMF playback and editing. If you use a keyboard workstation, make sure that it has a GM sound bank so commercial SMFs will play properly. You can always "de-GM" them later by deleting the first measure (GM setup) and assigning your own favorite sounds from other sound banks.

Whatever your performance needs may be, Standard MIDI Files can serve as a valuable part of your onstage experience. And of course, SMFs have many uses off stage, as well. To get a better idea of how SMFs have changed the lives of working musicians, let's look at some real-life scenes from that teeming jungle known as the music workplace.

### WHERE TO BUY STANDARD MIDI FILES

**Giebler Enterprises** tel. (610) 933-0332; fax (610) 933-0395
**MIDI Classics** tel. (800) 787-6434; Web www.midi-classics.com
**Midi Hits** tel. (800) 593-1228; fax (650) 637-9776; e-mail midihits@pacbell.net; Web www.midi-hits.com
**Norton Music & Fun** tel. (561) 467-2420; fax (561) 467-2420; Web members.aol.com/nortonmidi
**Peter Solley Productions** tel. (888) 211-0634; fax (954) 570-9788; Web www.petersolleyproductions.com
**Tran Tracks, Inc.** tel. (800) 473-0797 or (973) 383-6691; fax (973) 383-0797; Web www.trantracks.com
**Trycho Tunes** tel. (800) 543-8988 or (909) 696-3577; fax (909) 696-3571; e-mail trycho@mindspring.com; Web www.trycho.com
**The Works Music Productions** tel. (800) 531-5868; Web www.worksmidi.com

### SCENE 1: Keep It Casual

The casual scene in Los Angeles is a never-ending pickup gig with an ever-changing lineup of players on stage. It's not unusual to see a band performing splendidly and find out later that some of the musicians just met that night. To make sure everything works smoothly, some well-organized bandleaders provide their players with arrangements of songs that they're expected to know; and they use Standard MIDI Files to do it. With SMFs, it's easy to solo or mix one part above the rest as a tutorial for a particular player. For musicians who don't have MIDI gear, the bandleader simply records the MIDI playback onto a cassette tape. If everyone learns their parts, rehearsals can be kept to an absolute minimum.

The bandleader makes the necessary phone calls and gets several wedding dates matched up to the available players. Unfortunately, it's not always a perfect match. It looks like an upcoming gig is in jeopardy because the first-choice bass player is sick, and his replacement is unavailable. The drummer is *not* thrilled about that.

The solution: bring in a Standard MIDI File player and a GM synth module to play the bass parts. And while you're at it, break out the disks with the string and brass parts and really live it up. The bandleader has obviously done his homework by taking presequenced SMFs of the songs in the band's repertoire and editing them on his computer for key, tempo, mix, and click track. The wedding party eats it up (along with the wedding cake), and of course, one less band member means there's more money for everyone else. The drummer *is* thrilled about that.


**FIG. 2: Yamaha's portable, battery-powered MDF3 (MIDI Data Filer) reads and stores Standard MIDI Files (formats 0 and 1) on 3.5-inch floppy disks.**

### SCENE 2: Horns of Plenty

Two horn players got caught in a downsizing squeeze when the owner of the club where they worked decided that he was paying too much for entertainment. They had been part of a seven-piece band that smoked the Bay Area crowd five nights a week. These guys were amazing musicians, but they were reluctantly replaced by sequenced tracks from Standard MIDI Files. The band, it seems, saved the gig by having two fewer players to pay. Everyone agreed that this was a brutal economic reality, and sadness fell over San Francisco.

Well, the two horn players went out and got the necessary MIDI gear and a collection of SMFs. They then put together the same performance but in the opposite way. Instead of muting out the rhythm section, as the guys in their old band had, they muted out the horn parts and used the drums, bass, and other backing parts as needed. And because both of them could sing, dance, play an instrument, and work a crowd, they were ready to climb right back on stage after a couple of months of retooling. Best of all, about a year later, they were playing in the same club again, making way more money than they had before.

### SCENE 3: In the Pocket

I get calls all the time from singers who have lost their gig or lost their band and want to resume performing using Standard MIDI File backing tracks. These clients generally don't have MIDI gear, so they get an assortment of MIDI files and take them to a friend's project studio. They don't need anything fancy, just a decent sequencer and a good GM sound module. After a couple of passes through the tune for auditioning, they're ready to mix down. They simply mute out the melody line, transpose the key as needed, adjust the tempo a little, and then record their

For manufacturer information, please see Contact Sheet, p. 227.

MIDI tracks as digital audio onto a Sony MiniDisc.

Moreover, with the expanded professional 4-track MiniDisc units from companies such as Tascam and Yamaha, you can put the drums, bass, music mix, and background vocals on separate tracks. They would then be all ready for mixing and equalizing from the main board at the performance venue. You'd be surprised how many people are taking high-quality digital recordings of MIDI files to their gigs and performing over them. With the instant access to songs that the MiniDisc format provides, it's a surefire winner, and you can carry your band around in your pocket.

## SCENE 4: Net Results

The Standard MIDI File has long been the unloved cousin of digital audio. A pitiful limit on channels, sounds, and polyphony—not to mention unpredictable playback and lack of vocals—have all conspired to hamstring its effectiveness. But look! Up on the Net! It's a bird...it's a plane...it's a MIDI file! Those midget-sized files can play some big music in real time right through your Web browser. And with a premium software synth (such as Roland *Virtual Sound Canvas*, Yamaha *MIDPlug*, or Apple's QuickTime Musical Instruments) and some decent multimedia speakers (with a subwoofer), they actually sound pretty good.

You don't have to wait forever to hear a compressed-to-death WAV file or put up with the inevitable stuttering of low-bandwidth streaming audio. Sure, the MIDI file doesn't include vocal parts or the custom samples used on the original recordings. But the sound coming out of a high-quality sound module is surprisingly good, and the performance can stream in real time through LiveUpdate's *Crescendo* plug-in. (For more on this topic, see "Desktop Musician: Streaming MIDI" in June 1998 issue of **EM**.)

Only MIDI files can offer so much music with so little bandwidth. And it gets better: RealNetworks has developed a version of its RealAudio streaming technology that works with a new *Crescendo* plug-in to provide simultaneous (and synched) audio and MIDI. Now we're talking! The effective real-time bandwidth can be devoted almost entirely to a vocal performance, while the MIDI file, easy on the bandwidth, plays the backing track. No high sampling rates are needed for the mostly mid-range human voice. It's a powerful match and a fantastic opportunity for Web commercials using music with voice-overs. A demo of this technology at a recent developer's conference blew me away. You can experience it for yourself at www.liveupdate.com. (Find your way to the Forte showcase. As always, be prepared to download a lot of stuff to make it all work—it's worth it.)

## SCENE 5: Karaoke Kicks

It's 11:00 P.M. in a karaoke bar in Tokyo, and things are cooking. People are lined up to live out their fantasies on the big stage. There's only one problem. A nice young man wants to sing "Zoot Suit Riot." But the karaoke DJ doesn't have "Zoot Suit Riot," since it's such a new song. The DJ, however, is a master of his trade, and he quickly logs on to his favorite SMF vendor's Web site on the Internet. He hits the search engine and the tune he's looking for, and in a few moments, the SMF is transferred to his computer. He then loads it into his Standard MIDI File player, mutes the melody line, lowers the key, and speeds up the tempo a bit. In a matter of minutes, the backing track is pumpin', the club is thumpin', and another fantasy is fulfilled.

Later that night, the DJ has a brain boost: why not go to that Web site every morning and download everything that's new? That way the latest SMFs will already be on his hard drive when he needs them. And while he's at it, he could check other commercial Web sites for titles that people have been asking for. Ah, they've got the new Gloria Estefan tune; better order that one now. Click!

## GOING SHOPPING

When shopping for Standard MIDI Files, the most important considerations are how musical and how usable the files are. Take the time to assess how accurate and complete the arrangements are. (Some Web sites offer demos of their songs, so you can try before you buy. With the right plug-in, you can even audition the demos in real time.) Are the guitar or sax solos on separate channels for easy muting? Are the tracks laid out logically? Are you going to be up all night tweaking the file to make it work with your gear? Does the arrangement groove? If it does, can you groove along with it? And don't forget the ending: no fades, please.

The main point to remember is that commercial SMFs, especially when used on stage, still need a level of human involvement to be effective as entertainment. No one is going to be moved just listening to a MIDI file. As a performer, you've got to get out in front of a backing track so the audience can see you, hear you, and feel your presence. People love a good song regardless of how it's created. If good music incorporates a synth being ruled by an 8-bit stream of ones and zeroes, who cares? If the audience is entertained, and everyone goes home happy, that sounds good to me.

**Tran Whitley**, *president of Tran Tracks, has been in his studio for so long that exposure to direct sunlight could be fatal. He can be reached by e-mail at trantracks@crystal.palace.net.*

The songwriter/composer's dreams have just come true. Large power. Less space. All the lush sounds of EMU®. Twice the polyphony of the competition – 128 voices! And much more. So, go for it!

# PLAY LARGE

with the all new

## PROTEUS 2000

Play large with up to 128 MB of sounds, using Proteus® 2000's four 32 MB ROM slots. With the included 32 MB Composer sound-set, you get a colossal 1024 presets, 512 user-preset locations, and the room to grow to a massive 128 MB of ROM! All the sounds you need for your full orchestral score or your next hip-hop groove.

Tame this massive number of presets with SoundNavigator; instant access to every sound you need. Exploit all 32 ultra-fast response MIDI channels to get tight-tight grooves and near limitless sequencing with the Proteus 2000's 32-bit processor. Instantly save and recall your Multimode setups, so all 32 channels of your preset, volume, and panning data is one click away. Use 12 real-time controls (three banks, four knobs) to tweak your sounds instantly without touching the edit buttons.

And there's much more...digital output and dual 24-bit FX processors. Deadly EMU filters. Downloadable operating system and presets so you can conveniently take advantage of further EMU innovation. Never before has there been so much power in a single rack space sound module. Proteus 2000 allows you to take advantage of all the latest sounds, technology, and features EMU has to offer, so play on and PLAY LARGE with the Proteus 2000 sound module – the millennium won't wait.

128-Voice Polyphony – for creating lush sequences and massive layers without dropping a note.

Expandable – up to 128 MB of ROM presets in four 32 MB slots and comes off the shelf with the 32 MB Composer ROM installed – that's an incredible 1024 presets on board out of the box.

SoundNavigator – puts all your pianos in one room by giving you immediate access by bank, instrument category, or preset name to every sound in the box.

Roll Your Own – Coming soon, create your own Sound-Sets using EMU's newest authoring tools for unlimited sound capabilities - vocals, drum loops, effects, string quartets, and your old synths all integrated into your Proteus 2000...

For more information, visit out website at:
**www.emu.com**

**$995**




*East Coast Office*
155 Great Valley Parkway
P.O. Box 3035
Malvern, PA 19355-0735
(610) 647-3930

*West Coast Office*
1600 Green Hills Road
P.O. Box 660015
Scotts Valley, CA 95067-0015
(831) 438-1921

**E-MU SYSTEMS**

circle #553 on reader service card

SEE US AT NAMM BOOTH #W1233

RECORDING MUSICIAN

# Recording Resonator Guitars

***Capture stunning sounds from an American classic.***

By Brian Knave

Resonator guitars have made quite a comeback in recent years. This fact can be attributed not only to the revived popularity of acoustic instruments in pop and "alternative" music but also to the number of hot new players invigorating the scene: Corey Harris, Sonny Landreth, Alvin Youngblood Hart, Chris Whitley, and Rob Ickes, for example. It helps, too, that several high-profile artists, like Bonnie Raitt, John Fogerty, Mark Knopfler, and Joe Perry, play resonator guitars. Of course, one also has to credit the "veteran" stylists—guys such as Bob Brozman, Mike Auldridge, Taj Majal, John Hammond, and Leon Redbone—who kept the resonator torch burning throughout the '60s, '70s, and '80s while most other guitarists were focused elsewhere.

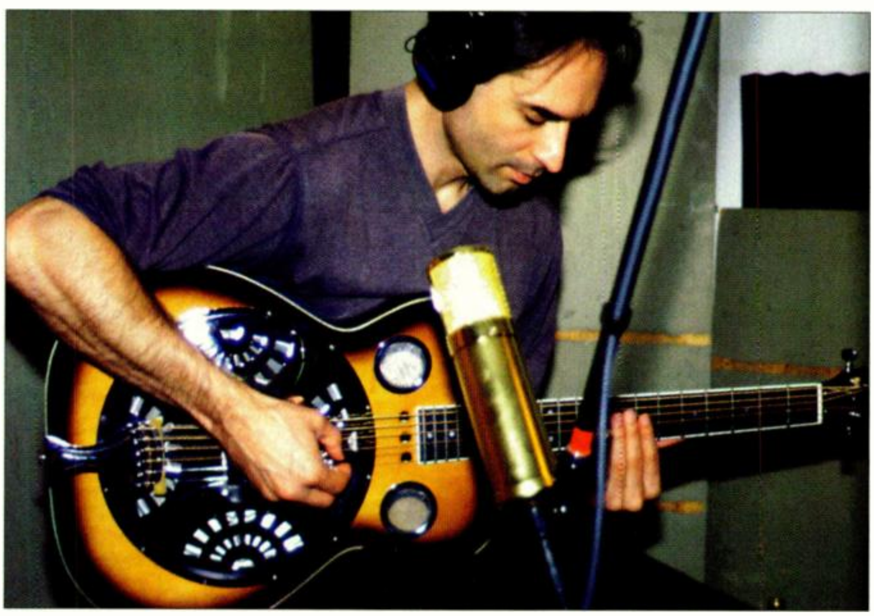

BRIAN KNAVE

**FIG. 1: A Lawson L47MP tube mic is positioned to capture a balanced mono image of Carl Weingarten's Regal Standard single-resonator guitar.**

Another factor in the recent revival of the instrument is the reappearance of National guitars, which were out of production for more than 40 years. (For more information on the evolution of resonator guitars, see "A Tale of Two Brothers" on the **EM** Web site at www.emusician.com.) Taken together, these developments increase the likelihood that you will be called upon to record a resonator guitar sometime in your personal-studio career.

So, you're thinking, what could be easier? After all, resonator guitars put out a heck of a lot of sound. Can't you pretty much just stick a mic anywhere near one and capture a usable sound?

Actually, recording this type of instrument is more complicated than that. For one thing, there are several varieties of resonator guitars, and understanding the different types and features is helpful to the recordist. Also, the playing styles that are used vary a good deal (for example, fingers vs. fingerpicks and bottleneck vs. chording) and must be taken into account when recorded. And then there are the usual considerations of room acoustics,

mic selection, mic placement, and so on.

I recently interviewed Daniel Thomas and Bil VornDick, two experts on the subject of recording resonator guitars. Thomas, chief engineer at Solo Sound Productions in Santa Cruz, California, has engineered all of Bob Brozman's records for the past seven years. VornDick, who was originally chief engineer for country music legend Marty Robbins, has recorded numerous resonator guitar artists, including Leon Redbone, Jerry Douglas, Rob Ickes, and Sonny Landreth. He also recorded and mixed the Grammy-winning album *The Great Dobro Sessions* (Sugar Hill Records, 1994), produced by Jerry Douglas and Tut Tayor.

I spent several days in the studio experimenting with Thomas and VornDick's techniques while my assistant, Carl Weingarten, was laying down tracks for a new album. Weingarten, a composer, slide guitarist, and resonator guitar aficionado, has recorded for various labels, and his music can frequently be heard on the syndicated APR program "Echoes" and on National Public Radio's "All Things Considered." Weingarten is also Webmaster at Planet Dobro, a site devoted to the culture of slide guitar (www.mphase.com/planetdb.htm).

These interviews and sessions have yielded the tips and techniques in this article, which will prepare you for recording resonator guitars.

## BRIDGES, CONES, AND TONES

We'll start by looking at the features of resonator guitars that are responsible for their tonal variations. Basically, there are two types of body materials—metal and wood—and two types of resonator systems—single cone and tricone (containing three small cones).

A tricone resonator has a T-shaped aluminum bridge that connects to the centers of three 6-inch cones. A wooden saddle sits on top of the T-bridge. Because vibrations must travel through the saddle to the T-bridge and then to the three cones, tricones have a quieter attack than single-resonator guitars. Overall, they sound smoother and more sophisticated, with greater sustain and more complex harmonic content.

Single-cone guitars come with either 9½-inch or 10½-inch cones (the larger cones are louder and produce more low end) and one of two types of bridges: a *biscuit* or a *spider*. With the biscuit bridge, the strings rest directly over a wooden, biscuit-shaped saddle that drives a convex cone. In the spider style, the strings rest across a spider-shaped assembly over a concave cone, vibrating the cone from the edges rather than from the center. Single-cone guitars often have a more banjo-like sound and a faster decay than tricones. Of the two bridge styles, the biscuit bridge produces a thinner, more muted and trebley tone—the sound that is often preferred by musicians who play Delta blues.

As for body material, steel-bodied guitars tend to have a punchy tone with a somewhat "clangy" resonance—ideal for a Son House or Charlie Patton kind of sound. Brass bodies, on the other hand, produce a smoother, clearer, and more controlled sound with a richer harmonic structure; these types of guitars are often favored for ragtime, jazz, and swing, due to their more articulate, refined sound. Wooden bodies produce a warmer, darker, and less sustained tone than metal bodies and typically are preferred for bluegrass and country music.

## NECKS AND SLIDES

Resonator guitars can have square or round necks. The square neck allows for lap playing only, whereas the round neck can be played either in the lap or upright like a standard guitar. Obviously, the two styles require different mic positioning.

I recorded two resonator guitars during the sessions, both played upright: Weingarten's Regal Standard, a wood-body, single-resonator guitar with a spider bridge; and a National Reso-Phonic Style 1 tricone with a nickel-plated brass body.

Weingarten, who plays finger style sans fingerpicks, played the guitars with slides made of brass, chrome, ceramic, and glass. The metal slides provided a fatter tone with more sustain as well as more string and fret noise; the ceramic and glass slides sounded thinner and less noisy. Of the metal slides, the brass one produced the twangier tone, and the chrome one (which was more polished than the brass) created more sustain. Of the other two, the ceramic sounded smoother and fatter than the glass.

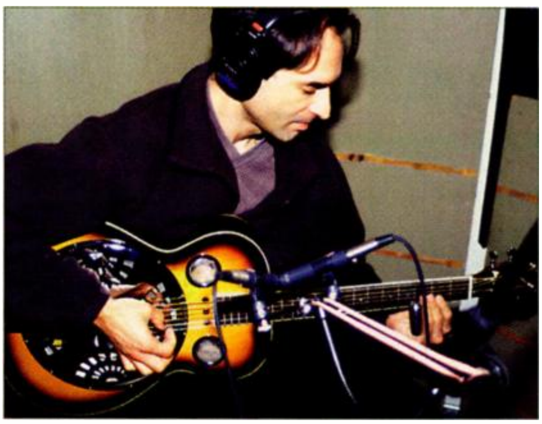
## BEFORE THE SESSION

Resonator guitars can be finicky instruments: sometimes the metal cones develop rattles or sympathetic buzzes at certain frequencies. Hopefully, players will take care of any unwanted noises before the session begins. If they don't, you may get sidetracked trying to squelch a buzz or rattle. You might have to alter microphone positioning to minimize offensive noises, and as a last resort, you may have to live with a bit of noise as part of the overall sound.

## TUNING THE ROOM

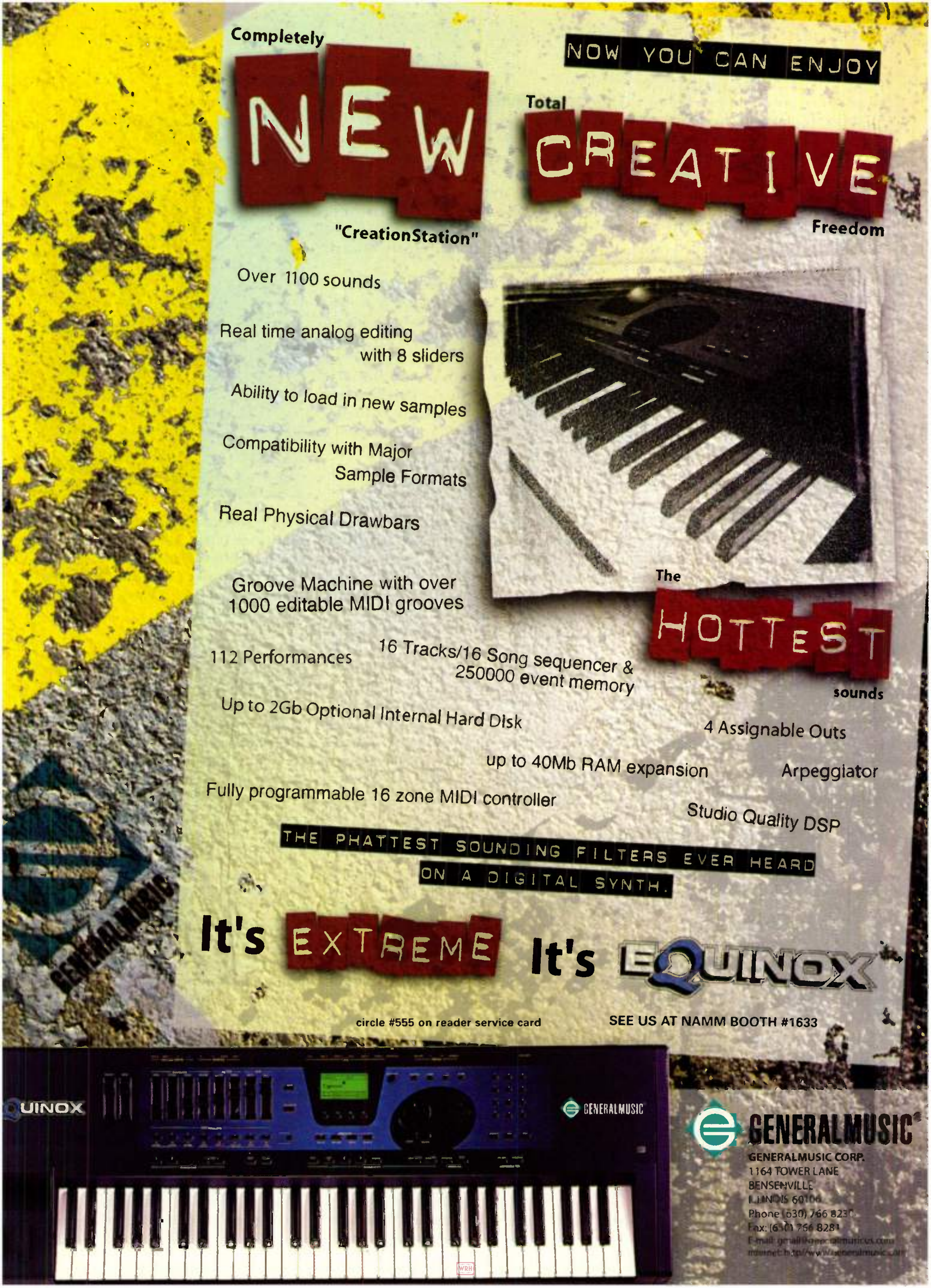
Not surprisingly, the sound of the recording space plays a significant role in the quality of the recording. Thomas and VornDick both stressed the importance of "tuning" the room to the track using wood panels (or some other reflective material) to increase early reflections and using baffles or gobos (fabric-covered office partitions work well) to dampen them. Both engineers prefer a "medium-dead" space—that is, one that is sufficiently live but not overly reflective.

Ideally, the room should have a wood floor. "If there isn't one," says Thomas, "I'll put down a large piece of wood—preferably oak with a glossy finish—or Plexiglas (which is lighter and easier to carry around) to create a reflective surface beneath the player." If the space is still too dead, try putting reflective panels behind and to both sides



BRIAN KNAVE

**FIG. 2: Coincident-pair stereo-miking is good for capturing a realistic, although narrow, stereo image. This pair of Neumann KM 184 cardioid condensers sounded great on Weingarten's resonator guitar.**


Completely
NEW
"CreationStation"
NOW YOU CAN ENJOY
Total
CREATIVE
Freedom
Over 1100 sounds
Real time analog editing with 8 sliders
Ability to load in new samples
Compatibility with Major Sample Formats
Real Physical Drawbars
Groove Machine with over 1000 editable MIDI grooves
112 Performances
16 Tracks/16 Song sequencer & 250000 event memory
The
HOTTEST
sounds
Up to 2Gb Optional Internal Hard Disk
4 Assignable Outs
up to 40Mb RAM expansion
Arpeggiator
Fully programmable 16 zone MIDI controller
Studio Quality DSP
THE PHATTEST SOUNDING FILTERS EVER HEARD ON A DIGITAL SYNTH.
It's EXTREME It's EQUINOX
circle #555 on reader service card
SEE US AT NAMM BOOTH #1633
GENERALMUSIC
GENERALMUSIC CORP.
1164 TOWER LANE
BENSENVILLE
ILLINOIS 60106
Phone (630) 766 8230
Fax: (630) 766 8281

of the player, and be sure to angle them so that they are not parallel to the mics. (**Figure 3** shows a dead room that has been made more live through the use of wood panels.)

Another workaround for an overly dead space is to mic closer to the instrument using a cardioid microphone that has a fairly controlled proximity effect. "On a tricone," explains Thomas, "try using an AKG C 460 right down next to the cone—the brightest part of the guitar—and bring the mic in close enough to get some bass boost. That's one of the cool things about the C 460 with the CK61 (cardioid) capsule: it has an incredibly controlled proximity effect, so that changing the angle is equivalent to changing the sweep control on your EQ. You just find the sweet spot that you're looking for.

"Next," continues Thomas, "send that signal to an early reflection reverb to simulate the 'bark' of a bright room. The trick is that you're sending only one signal to the processor; the other signal, which is from the mic on the sound hole, is left dry." In a pinch, Thomas has also obtained decent results with a Shure SM57 positioned to create the bass boost.

Of course, sometimes a space is too live, in which case you may need to position gobos around the musician to cut down on reflections. Gobos are also helpful if you are recording several musicians at once.

Be aware, too, that reflections from a nearby music stand can spoil your sound, so dampen them using a piece of carpet or cloth. It also helps to position the music stand off-axis to the mic and nonparallel to the surface of the guitar.

## MICS OF CHOICE

Selecting and placing your mics are the next critical steps. The first consideration is whether to mic stereo or mono. Both Thomas and VornDick prefer stereo miking, although for a song containing multiple resonator-guitar tracks, Thomas usually stereo-mics the lead voices and mono-mics the rhythm tracks. "Brozman often stacks up a lot of guitars on a recording," he explains, "using up to five or six rhythm tracks for one song. Usually, the rhythm tracks are played on a single resonator, so I'll mono-mic those to get a tight, contained sound."

Thomas likes using Brüel & Kjaer 4006 omnidirectional condenser mics for stereo recording and a B&K 4009 cardioid condenser for mono-miking, assuming that the room is well "tuned" and that he is going for a natural sound. He also likes Schoepps mics and the Neumann KM 150, especially for live recording, because the KM 150 is hypercardioid and rejects off-axis sound well.

VornDick, on the other hand, prefers using vintage tube mics such as AKG C 12s, Telefunken 251s, and Neumann U 67s. "Resonator guitars are pretty bright and brash sounding," he explains, "so the tube mics help attenuate that from the get-go. For the metal bodies, I would lean toward tube or Class A mic preamps, as well. You're still going to capture the brightness and brashness, but it will be smoother."

In a personal studio, where such high-end mics aren't likely to be found, good substitutions for VornDick's suggested tube mics would be the AKG SolidTube, GT Electronics AM61 or AM62, and Lawson L47MP (from which Weingarten and I got excellent results). As a substitute for the omni B&Ks, try any of the Earthworks omni mics (TC30K, TC40K, and QTC1); for the cardioids, try Earthworks Z30s.

Thomas also likes large-diaphragm condenser mics for mono-miking. "For the personal-studio audience, the Audio-Technica AT4033 is a good choice," he says. (VornDick also recommended the AT4033.) "It's a bit too bright, but you can dampen it a little with EQ or, as is my preference, turn it a little bit off-axis. If you get too much of the room when turning it off-axis, then dampen the room some more. Distance would be anywhere from one and one-half feet to three feet, depending on how much room sound you want. If the musician is playing upright, then place the mic so that it aims between the cone and the sound holes, and then play with the angle to control the brightness or change the EQ." (See **Fig. 1** for an example of this approach to mono-miking.)

## STEREO STYLING

VornDick, who always uses two microphones to record a resonator guitar, generally prefers the spaced-pair technique. "I try to capture the sound the way the artist is hearing it," he says. "If the guy is playing the instrument upright, like a regular guitar, I'll move the mic on the resonator side behind the instrument a bit, back from the right shoulder—or even behind the head, close to the ear. Of course, it also depends on the environment, whether it's a live or dead room. But basically, I try to paint a picture of the sound in the environment where it is occurring.

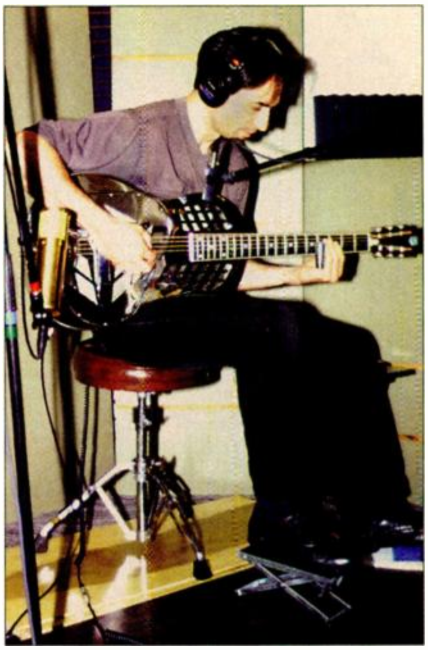

BRIAN KNAVE

**FIG. 3: In this spaced-pair stereo-miking setup, the Lawson L47MP hears the resonator and room tone while the Neumann KM 184 captures the more bassy tones from the sound holes. Note the wood panels arranged to liven up the sound with early reflections.**

"The other mic," he continues, "is down near the 12th fret. In the case of a metal body, I would move the mic off-axis a bit to keep the noise down and maybe come in closer to where the neck hits the body."

VornDick has also gotten great results with a coincident pair. (In a coincident pair, also called an XY, the mic capsules are as close together as possible, so the sound reaches each at the same time; typically, the mics are angled at 90 degrees to one another.) "I tend to place the XY about three to six inches down from the chin of the player," he explains, "between the chin and the body of the guitar, pointing down at the guitar between the bridge and the body. With an XY, you're getting that outside area of the guitar, which helps to pick up the higher overtones."

Weingarten and I experimented with XY-coincident and spaced-pair stereo-miking and got excellent results from

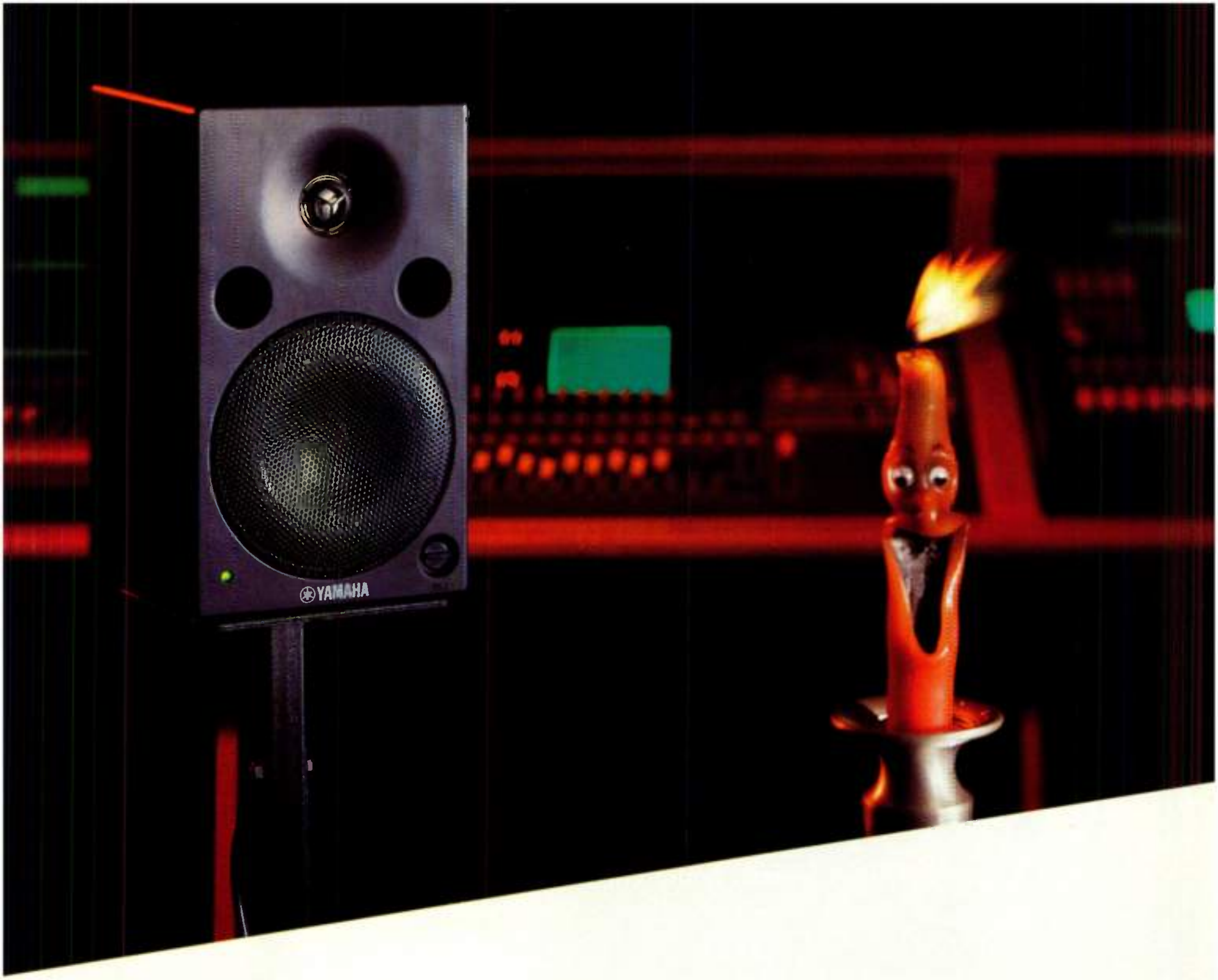

YAMAHA


FIG. 4: We experimented with ribbon mics, in this case a Royer Labs R-121. The final position of the ribbon mic ended up being about two inches back and an inch or so higher than shown here, with more of a downward angle.

both. For the XY-coincident tracks, we used a matched pair of Neumann KM 184s (see **Fig. 2**). For the spaced-pair tracks, we also tried the KM 184s, but we got better results using nonmatched microphones: for example, the Lawson L47MP and one KM 184 (see **Fig. 3**). Overall, we preferred the sound captured by the nonmatched spaced pair because it offered the most tonal variation—a nice thing to have when it comes time to mix. However, the XY-coincident tracks provided a more realistic "picture" of the sound.

If the resonator guitar has a pickup and you have tracks to spare, it's not a bad idea to dedicate a third track to the pickup's signal, if only to allow for more options at mixdown. Also, this signal can be helpful for adding definition to bass notes, especially in the case of a walking bass line. The National Reso-Phonic tricone that we miked contained an exceptionally good-sounding Highlander pickup, so we routed that signal to its own track.

## BARK AND CREAM

When miking tricones, Thomas goes for one of two sounds: "bark" and "cream." To get what Brozman calls bark, I position both mics seven to eight inches from the instrument. The mic on the sound hole is at a 45-degree angle to the hole (or holes), where your bass response comes from. If you get too close to that area, it will sound awful. But off-axis, seven to ten inches out, you get a really nice bass sound.

"With a tricone, first listen to all three cones, and find the one that sounds best and isn't buzzing. Start out by positioning the mic to aim at the area between the center and edge of that cone, just as you would when miking a speaker in a speaker cabinet. Then play with proximity effect. If I'm not getting the bass response that I want off of the sound-hole mic, I go with proximity effect on the other mic and bring it in close to the cone. Then, using either mic placement (preferable) or an outboard EQ, I might notch those frequencies from the sound-hole mic that are being *boosted* on the cone mic. That way, the output of the two mics dovetails, and the sounds fit together better."

Thomas also recommends miking the lower sound holes rather than the upper ones, so as to capture less breathing from the artist. However, on the tricone that Weingarten and I recorded, I got a better sound from the upper holes, so I diminished breath pickup by tilting the mic downward.

"'Bark,'" continues Thomas, "is a whole lot of 1.8 kHz. If I go above that, Bob says, 'Wrong sound'; if I go below it, he says, 'Too thuppy.' That's the Brozman sound: 1.8 kHz boosted. I try to do it with mic placement, but if we're on a tight budget and need to move fast, I'll do it with EQ, which I wouldn't recommend to anyone."

To get a "cream" sound, Thomas brings up the level of the sound-hole mic and reduces the level of the mic on the cone side. Also, he sometimes moves the cone-side mic away from the guitar a bit to pick up more of the body and less of the cone. This reduces the sound of the slide moving on the neck and enhances the fundamental of the note. "As the bark factor goes down, you start getting a lot of 500 and 750 Hz in the blend," explains Thomas. "This brings out the fundamental, resulting in a very smooth, creamy sound with lots of sustain. That's the sound I use for lead passages."

## MIKING LAP STYLE

Wood-body Dobros are often played lap style, especially by bluegrass and country musicians, and this calls for different mic positioning and perhaps different mics. "With wooden bodies," says VornDick, "you can lean more toward solid-state mics. Ribbon mics, such as the old RCA 77BX, also work well." (See **Figure 4** for a shot of a ribbon mic in action.)

"Usually, I have one mic—a Neumann U 67 FET, for example—shooting from overhead at about a 45-degree angle, aimed between the body and the player's playing hand. This not only brings down the brashness, but also prevents any problem with the parallel diaphragms (the mic's and the Dobro's) resonating against each other, which can cause a weird overtone distortion. If there are other live instruments on the session, I put the Dobro's mic about 6 to 12 inches away from it; the more instruments there are, the closer the mic.

"The second mic is positioned where the neck meets the body," continues VornDick, "kind of near the hole but slightly off of the guitar, at a 45-degree angle. That way, I get a little bit of the slide, but not too much. Also, that positioning helps cut down a bit on the fast overtone transients of the bar sound. For this mic, I like to use a Neumann KM 84 or a U 47 FET. This approach to stereo-miking, which is essentially dual mono, allows me to mix the two sounds the way I want later."

VornDick positions the two mics anywhere from 12 to 18 inches apart, taking care to listen for phase problems. He also stresses the importance of making sure the mics themselves are in phase with one another, especially when using vintage mics, as some older models were wired with pin 3 hot. As for using lower-end mics, VornDick says he has gotten decent results using either CAD/Equitech E200s or Audio-Technica AT 4031s.

## EQ AND PANNING

Both VornDick and Thomas prefer to avoid EQ when possible, but there are exceptions to every rule. "When I mix," says VornDick, "my approach is to subtract frequencies rather than boost. For example, I might take out some upper mids between 2 and 5 kHz to pull back the 'ienk' sound of a metal-body instrument. But it also depends on what key the song is in and where the problem lies.

"Sometimes, if you take off a little of the high end, especially on the Nationals, it tightens up the sound and adds a little more beef," continues VornDick. "Or, if you want to fatten up the sound, you can boost a little between 150 to 300 cycles. Just be conscious of which other instruments are generating frequencies in that range."

Whether your stereo tracks were recorded with a spaced pair of mics or with an XY setup, be careful not to pan them too far apart. With a spaced pair, a hard pan will put the lower frequencies on one side and the higher frequencies on the other; with XY tracks, a hard pan will spoil the integrity of the image. Either way, the results will sound unnatural. Instead, go for a soft pan: for example, put one track at 10 o'clock and the other at 2.

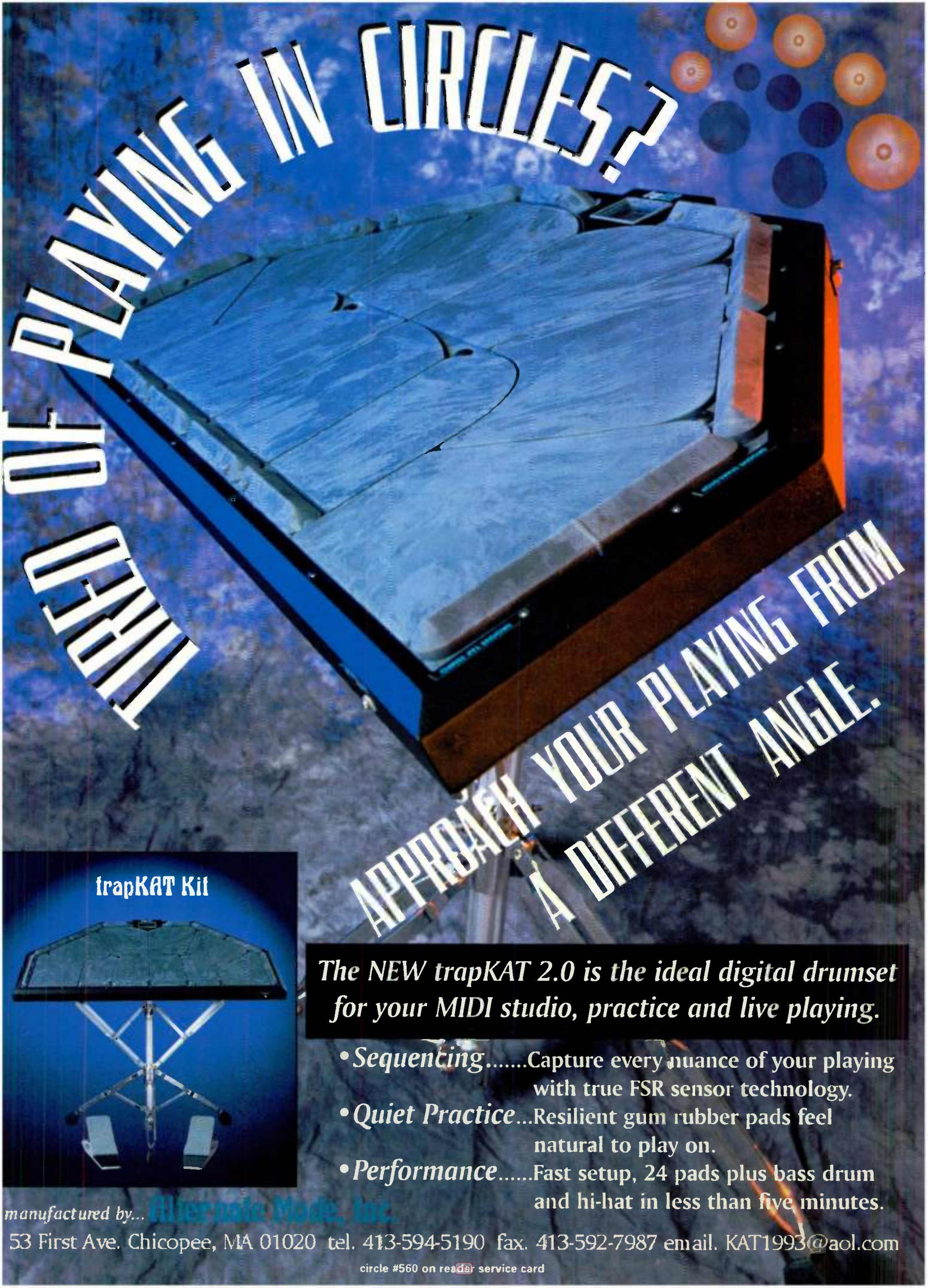
## BLIND TRUST

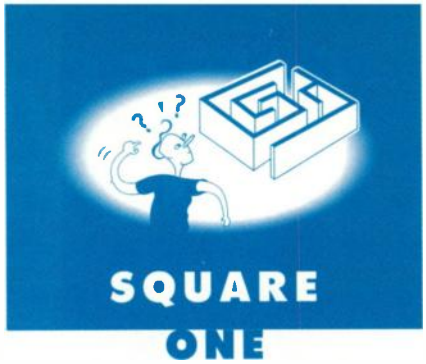
Obviously, every instrument sounds unique and every room has its own acoustics, so your ear must be the final arbiter as to which mics you use and how you position them. Moreover, you should regard the recommendations offered here as guidelines only, not rules.

That said, I will leave you with a final tip, one I heard years ago in a ceramics class while learning to throw pots; nevertheless, it applies equally well to positioning mics. Before making a pot, you must center the clay on the wheel. This is more difficult than it looks. In fact, I struggled with it for days. Then, one evening, as I labored over a mound of spinning clay, a veteran potter said, "Close your eyes." I did, and I could immediately *feel* the balance of the clay in my hands. In moments, the lump of clay was perfectly centered.

The same goes for finding the sweet spot when positioning mics. Once you have everything set up and are ready to finalize a mic's position, close your eyes as you move the mic around. That way, you shut off visual cues (including all the notions you've gotten about mic placement from looking at photographs and diagrams), which leaves you with one thing only: the sound. And that's all you need in order to find the mic position that sounds best. Indeed, you should rely on nothing else.

*Thanks to Myles Boisen, Bob Brozman, Mary Cosola, McGregor Gaines, Daniel Thomas, Bil VornDick, Carl Weingarten, and Todd Wright.*


TIRED OF PLAYING IN CIRCLES?
APPROACH YOUR PLAYING FROM A DIFFERENT ANGLE.
trapKAT Kit
The NEW trapKAT 2.0 is the ideal digital drumset for your MIDI studio, practice and live playing.
• Sequencing.......Capture every nuance of your playing with true FSR sensor technology.
• Quiet Practice...Resilient gum rubber pads feel natural to play on.
• Performance......Fast setup, 24 pads plus bass drum and hi-hat in less than five minutes.
manufactured by... Alternate Mode, Inc.
53 First Ave. Chicopee, MA 01020 tel. 413-594-5190 fax. 413-592-7987 email. KAT1993@aol.com
circle #560 on reader service card




# Get in Sync

***Some things to sync about when you need to connect your gear.***

By Dan Phillips

Synchronization—getting two or more devices to record or play back together—is a fundamental factor in making a studio work. Sync issues are similar for analog and digital devices. In fact, synching digital devices is often simpler than synching their analog counterparts. There are some different methods in the digital synchronization toolkit, which may be unfamiliar to you. In this article, we'll look at the basic concepts of synchronization and how they apply in the digital domain.


STEVE RAMIREZ

First, it's important to realize that synchronization has two components: one is starting at the same time, and the other is proceeding at the same speed. For the sake of convenience, we'll call the first "start sync" and the second "continuous sync." In the analog world, these are often two facets of a single synchronization method, such as SMPTE time code. With digital devices, on the other hand, they may be handled separately.

## ANALOG SYNC

Let's start with the analog case first. With analog tape, both start sync and continuous sync are generally handled by SMPTE (Society of Motion Picture and Television Engineers) time code. For instance, to synchronize two analog 24-track tape machines, you would stripe both tapes with SMPTE (that is, record time code onto a track) and connect the machines using an analog tape synchronizer.

The synchronizer listens to SMPTE from both machines and directly controls their motors, and also provides simple transport. When you press Play on the master machine, the synchronizer shuttles and varispeeds the tapes until both machines are playing back from the same point in the music: at the beginning of the second chorus, for example. Once this happens (and it can sometimes take a while), start sync is achieved.

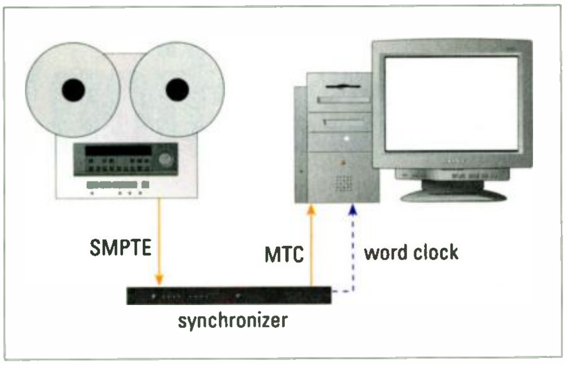



**FIG. 1: When synching a computer-based digital audio system to an analog tape deck, a synchronizer translates SMPTE from the analog deck into word clock for the computer hardware.**

If that were all that the synchronizer did, the tapes would start together but then gradually drift apart. This is because the machines' motors run at slightly different speeds, even though both are ostensibly moving the tape at 30 inches per second. To solve this problem, the synchronizer listens to the two SMPTE streams throughout playback, adjusting the tape speed as necessary so that the machines keep playing together.

## ANALOG-TO-DIGITAL SYNC

Synching an analog system to a digital system, either tape or a DAW, is conceptually the same as synching two analog machines. The two systems need to start at the same point, and they must agree on the playback speed.

As with analog machines, time code, such as SMPTE or MIDI Time Code (MTC), is used for communicating the start point. Continuous sync, however, is achieved differently, because DAWs have no tape motors to control. Setting the playback speed must be accomplished with *word clock*.

A digital audio recording consists of a long stream of individual samples. You might picture each sample occupying a very small area on the surface of an analog tape. Just as analog tape moves across the tape heads at a constant speed (such as 30 inches per second), the samples within a digital audio recording flow by at a specific sample rate (such as 44.1 or 48 kHz).

The word clock controls this flow by setting the precise sample rate of the digital system. Each time the clock "ticks," the system sends or receives another sample. If the device's sample rate is 48 kHz, for instance, the clock will tick once each 48,000th of a second. The word clock thereby controls the "tape speed" of a digital system. Even on common digital tape systems, in fact, the word clock is what controls the speed of the physical tape transport; the tape speed is adjusted to match the word clock.

Word clock can be generated internally or received from an external source. Most digital audio connection formats, including S/PDIF, TDIF, AES/EBU, and ADAT Optical, include the word clock signal. Word clock can also be transmitted separately, without audio data.

To sync a digital system to an analog one, you need to translate the analog tape speed into the digital word clock. A number of synchronizers (such as the Mark of the Unicorn MIDI Timepiece AV, Opcode Studio 64 XTC, and Digidesign Universal Slave Driver) do exactly that: they read the SMPTE or MTC signal from the analog tape and use the rate of the time code to generate a word clock signal (see **Fig. 1**).

For instance, if SMPTE is running at 30 frames per second, and the audio sample rate is 48 kHz, each SMPTE frame corresponds to 1,600 ticks of the word clock. Synchronization ensures that when the SMPTE from the analog tape runs faster or slower, the word clock, and thus the "tape speed" of the digital audio system, speeds up or slows down accordingly.

Some digital audio software and hardware does this internally, without changing the audio hardware's word clock rate. Instead, they measure the incoming SMPTE or MTC stream, varispeed the audio in software, and then send out the processed audio at a constant sample rate.

## DIGITAL TO DIGITAL

As with the all-analog and hybrid analog/digital cases above, all-digital setups also use time code for start sync. Depending on the equipment, they may use the familiar MTC or SMPTE time codes or new proprietary, sample-accurate time codes (as discussed later).

Continuous sync, happily, is much easier to achieve with all-digital setups. You no longer need to directly control physical tape speeds or varispeed word clock based on SMPTE or MTC. Instead, you connect the word clock ports of all the devices, so that the units play and record at exactly the same sample rate; thus, all will have the same "tape speed."

You might think that, with the precision of digital equipment, 48 kHz would be the same from one machine to another, making word clock connections unnecessary. Unfortunately, this isn't the case. Just as analog tape decks differ on what 30 inches per second means, so digital devices interpret sample rates differently. With two machines set to 48 kHz, for example, one might really run at 47.998 kHz, while the other might run at 48.001 kHz.

The solution to this dilemma is a master-slave setup similar in concept to analog time-code configurations. The master sets the sample rate, and the slave(s) ignore their own internal clocks, using the master's clock instead. A proper word clock arrangement will ensure perfect, driftless, continuous sync between all connected devices.

You can think of each device's word clock as a mechanical gear, with every tooth in the gear representing a single sample. Spinning by themselves, the gears can move at any rate they like. But when two or more gears are fitted together, they move in lock-step precision. Every time one gear advances by a single tooth, the others turn exactly one tooth as well—no more, no less.

When the word clocks of digital audio devices are synchronized, they act in the same way. Each time the master plays a single sample, the slaves play one as well. If two devices are playing the same digital audio file, they will

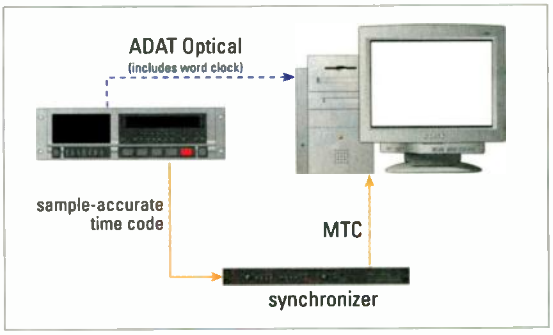



**FIG. 2: Synchronizers can translate an MDM's sample-accurate time code into standard MTC. In this case, the time code and word clock from the ADAT follow separate paths.**

take exactly the same amount of time to play that file.

It's important to note that word clock synchronization is critical not only for achieving continuous sync but also for transmitting digital audio data accurately under any circumstances. Without proper word clock sync, you'll be plagued by audio artifacts, including pops, clicks, and distortion.

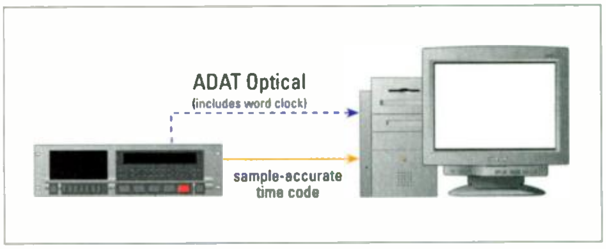



**FIG. 3: With hardware and software that directly support sample-accurate time code, you can connect the MDM directly to your computer hardware.**

As a side note, standard word clock signals tick once per sample. In contrast, Digidesign's systems use a "superclock," which ticks 256 times per sample. The two clocks are not directly compatible, although some synchronizers offer both options. Some may even offer a single set of word clock I/O that is switchable between the two, so make sure that you've selected the correct option for your gear.

## SETTING UP WORD CLOCK

Setting up a studio's word clock synchronization is usually done in two steps. First, the word clocks must be physically connected. You can do this by making a digital audio connection from the master to the slave, such as S/PDIF out to S/PDIF in, ADAT optical out to ADAT optical in, and so on.

Sometimes, you'll also need to use dedicated word clock cables, such as with complex setups or with devices that have word clock I/O but not digital audio I/O. With larger systems, you can also mix and match formats as required by the I/O available on each device.

The second step is configuring the slave devices to use the word clock from the master device. In digital audio software/hardware combinations, this is usually referred to as something like "sync source" or "audio clock source." On hardware devices, there may be an obvious front-panel selection (such as Clock Source on the ADAT XT), or a hidden key combination (as with original ADATs). Some older or simpler devices, such as DATs, may simply switch automatically as digital I/O is enabled or disabled, which can limit your options. Fortunately, most new devices are more flexible.

If your digital audio software supports software-based continuous sync, disable this feature; the word clock connection will take care of this via your hardware. In some cases, in fact, software-based continuous sync can interfere with hardware-based sync, causing audio artifacts or even total sync failure. Names for this parameter vary among software: in BIAS *Deck II*, for instance, you should enable Trigger Sync, while in Cakewalk *Pro Audio*, you should set the Audio Options>Advanced SMPTE/MTC Sync to Freewheel.

## THE FINAL FRONTIER

With a proper word clock setup in an all-digital system, it's easy to achieve perfect, sample-accurate continuous synchronization. The accuracy of the start point, on the other hand, depends upon the type of time code being used.

MTC, for instance, offers resolution of a quarter frame at 30 frames per second, or a 120th of a second—equivalent to about 400 samples per second at 48 kHz. SMPTE may offer greater resolution in some cases; however, if you're using a digital audio program, chances are that SMPTE must be converted to MTC before reaching the software, which means that you're still operating at MTC resolution.

ADAT and DA-88 time codes, in contrast, are based on the word clock. Each tick of the clock is also a tick of the time code, in a convergence of time code and "tape speed," which brings us full circle from the dual-function SMPTE of all-analog sync (see **Fig. 2**). Both systems offer single-sample accuracy for start sync, with a resolution of a 48,000th of a second at 48 kHz; hence the term *sample-accurate sync*.

Several combinations of computer-based audio software and hardware also offer direct support for sample-accurate time code (see **Fig. 3**). Due to the slight indeterminacy of computer operating systems, the sync may be perfect or it may vary slightly by a sample

# 32 Channels. Built-in MIDI. Solid sync. All together. Now.

## Introducing
## Dakota, Montana & Sierra

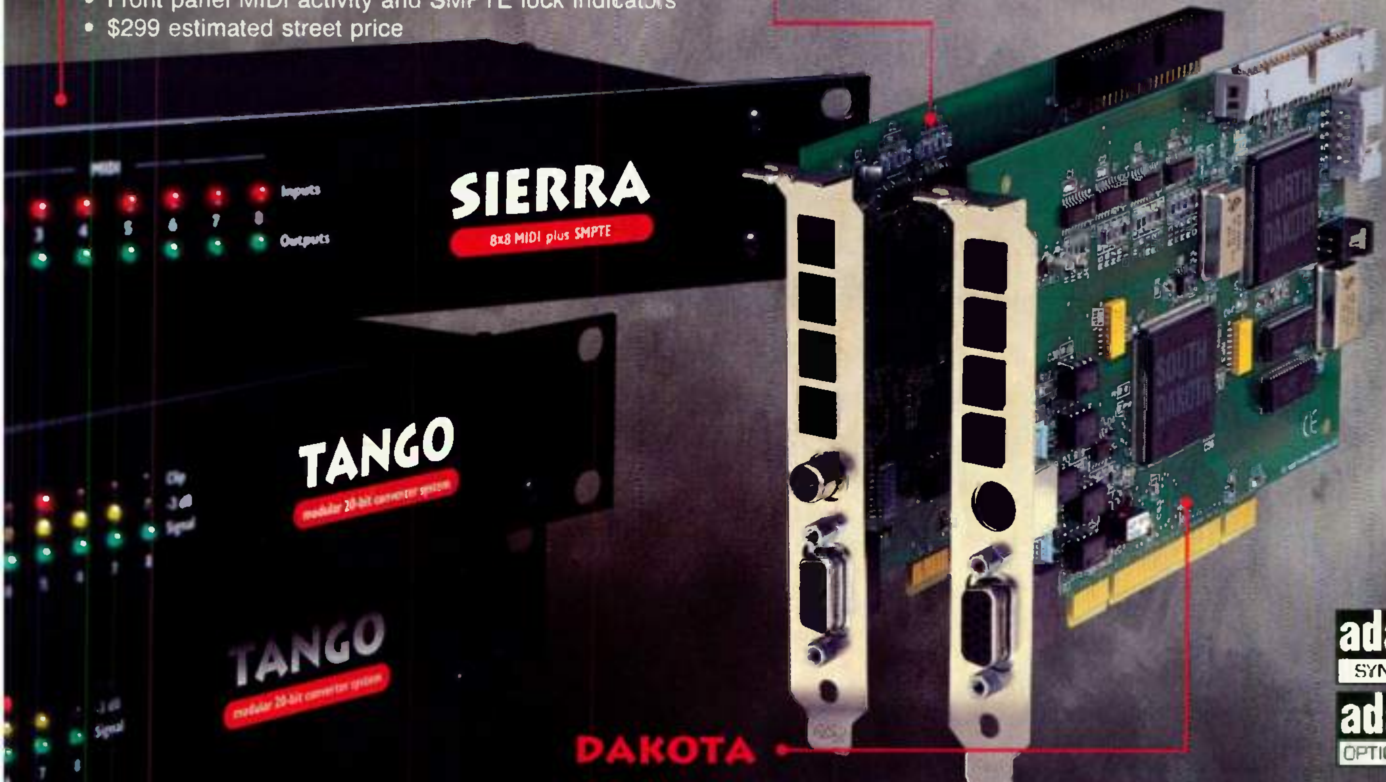
**The superintegrated digital audio, MIDI, and sync solution from Frontier Design Group**

### SIERRA

- Expands Dakota's MIDI I/O to 8 inputs and outputs
- "Quick-patch" I/O ports on front of unit
- 19" rackmount enclosure
- SMPTE input and output on 1/4" jacks
- Front panel MIDI activity and SMPTE lock indicators
- $299 estimated street price

### MONTANA

- Doubles Dakota's ADAT optical I/O to a full 32 in, 32 out
- Video sync/word clock input locks audio to standard NTSC/PAL video frame rates, or to external word clock
- ADAT Sync output
- Plugs into either PCI or ISA slot
- $249 estimated street price




adat SYNC
adat OPTICAL

### DAKOTA

- 16 channels of 24-bit ADAT lightpipe I/O
- 2 channels S/PDIF on coaxial or optical connectors
- Two low-latency MIDI inputs and outputs
- ADAT Sync input for sample-accurate synchronization
- Ultra-efficient bus-mastering PCI interface with scatter-gather DMA
- Exclusive: SoDA™ (SMPTE on Digital Audio) converts any digital audio channel to a SMPTE input or output
- True hardware chaselock to timecode
- Interfaces with all ADAT optical and S/PDIF gear, including Tango™ and Zulu™ external converters
- $649 estimated street price

**30-day money-back GUARANTEE**

VISA MasterCard

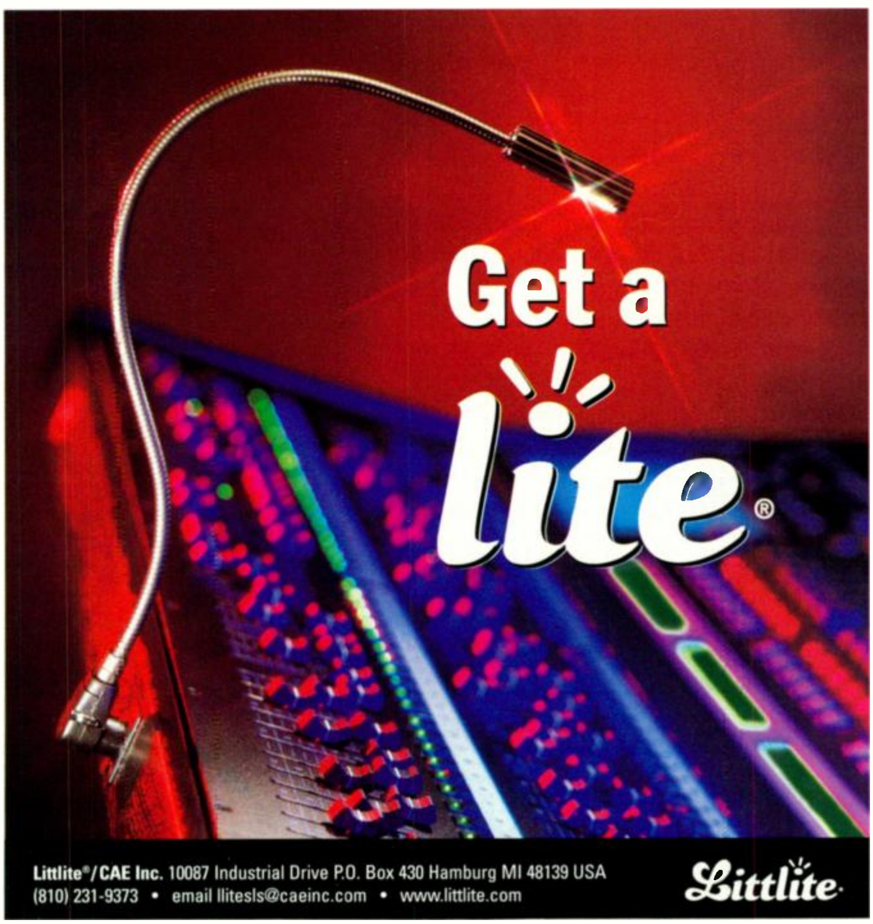
TO ORDER or FIND OUT MORE
**800.928.3236**
**www.frontierdesign.com**
**outside USA 603.448.6283**

**FRONTIER DESIGN GROUP**



circle #643 on reader service card

SEE US AT NAMM BOOTH #7016


or so. Even with a bit of slack, however, this is much more accurate than MTC.

## RESOLUTION AND RESULTS

So, what is the advantage of that increased start-point accuracy? The most compelling use is in computer editing of audio tracks from a digital tape recorder. With sample-accurate sync, the audio can be transferred into the computer and then laid back to tape with extreme precision.

To test the difference for yourself, record a track from an MDM into digital audio software, first using sample-accurate sync and then using MTC. Simultaneously play back both the computer and MDM tracks, again using sample-accurate sync for playback of the first and MTC for playback of the second, and compare the two. You'll notice that the sample-accurate tracks

**Think of word clock as a mechanical gear, with every tooth a single sample.**

sound louder, with little or no phasing, while the MTC tracks sound muted and noticeably phased.

Similarly, with MTC, tracks transferred in different record passes may have noticeable differences in phase. For best results when transferring phase-correlated audio (such as a drum kit recorded with multiple mics), it's best to transfer all tracks in a single pass.

Sample-accurate time code has certainly upped the ante in sample resolution. Remember, though, that almost every CD you own was recorded using SMPTE/MTC for synchronization (if the project used sync at all). They sound just fine, don't they? So, if SMPTE/MTC sync is all you have available, don't be too concerned.

Also, remember that regardless of start-point resolution, a proper word clock setup always ensures drift-free continuous sync. So, keep your eye on those word clocks, and be safe synchers.

---

**Dan Phillips** *is a singer, songwriter, and producer and is part of the team at Korg Research and Development.*

Pure Audio ...
24 Bit / 96 kHz
Electronic Musician
1999
EDITORS
CHOICE
Best Audio Quality
According to Bob Edwards, staff engineer at Lucasfilms Skywalker Sound; Baker Vaughn, mastering engineer at Tennessee Productions, Rainer Weber, mastering engineer at Nightwalker Studios, Germany.
High Performance Interface
Recording Magazine: "Best of the host based breed".
A wealth of effects
Multi-Band dynamics and stereo enhancer, declipper, denoiser, FFT draw filter 36,000 band EQ, room simulator and more at no additional cost.
The object editor
Apply EQ, dynamics, volume, pan, fade curves and stereo imaging surgically per region (in addition to any effects being used on the whole track).
CD Burning "On the Fly"
Master Redbook CD´s directly from a multitrack project, while processing effects and plug-in´s in real-time.
Total Compatibility
Doesn't lock you into a dead end system: By using the Intel CPU and any 95™ or NT™ compatible audio I/O devices, you are free to choose!
Samplitude 2496 – Red Roaster 5.0
Multi track audio recording, editing and CD-mastering software
SEK´D PRODIF and ARC audio I/O devices
Up to 24 Bit, 96 kHz digital, AES/EBU, S/PDIF coax and lightpipe, ADAT™ and analog I/O´s, wordclock, drivers for WINDOWS™ 95, 98 and NT (MAC and ASIO soon).
24/96 S
2496
Real 24-Bit 96kHz converters
ADDA 2496 S: Stereo ADDA. AES/EBU, S/PDIF, balanced analog I/O on XLR, 113 dB dynamic range, build-in power supply.
ADDA 2496 DSP: Adds word-clock, real-time resampling and dithering, records 24 bit 96 kHz audio on Dat tape, 128 dB dynamic range.
SEK'D
Digital Audio for the World!
circle #566 on reader service card
For more information and a dealer near you, please call
1 800 330 7753
SEK'D America
P.O.Box 5497
Santa Rosa, CA 95402
Tel: 707-578-2023
Fax: 707-578-2025
eMail: info@sekd.com
Web: www.sekd.com

# Taxing Your Talent

***Don't let bad tax planning ruin your day.***

By Michael A. Aczon

Being a musician is a taxing profession. News reports bring us grim reminders of superstar artists who find themselves hopelessly in debt for back taxes, simply because they did not plan correctly when stardom and money came along. This month's column is an introduction to some of the basic tax matters that you should think about when running your music business.

By far, the tax that gets the most attention is income tax, which is literally a tax on your income. The federal government uses the Internal Revenue Service to collect these taxes from you, and each state also has its own tax board for the same purpose. The money collected from income taxes is used to finance the day-to-day running of the government and the various programs it provides. Other taxes, such as sales and local taxes, can affect your career as well, but our focus will be on income taxes.

JACK DESROCHER

## WITHHOLDING AND DEPOSITS

When you began work as an employee in any kind of job outside of the music business, your employer probably had you fill out a form W-4 asking you various questions about your marital status and the number of children you have. You were asked these questions because employers must report to the IRS how much money you are being paid. Your employer withheld a portion of your total income during each pay period to approximate how much you should pay in income taxes to the government.

Any music business entity that pays you income—either as a salary or in royalties—is also required to withhold this portion of your income. That's why filling out a tax withholding form should be part of the signing process when you finalize a record deal or when you sign on as a side musician for a tour.

If those paying you for your services withhold too little of your income, you are responsible for approximating and paying to the government what you think you will owe in taxes at the end of the year. You do this by making quarterly deposits to the IRS and to your state taxing agency, if applicable. If you provide your services on a freelance basis, it is a good idea to calculate your taxes and make quarterly payments.

Let's suppose you are making steady money as a musician, and you are withholding or making quarterly payments based on your usual amount of work. If you make a record that becomes a hit, have a song used for a major motion picture, or are called to do a major summer tour, your income may rise dramatically. If it does, you should either adjust your withholding or make bigger quarterly deposits.

## DEDUCTIONS

Musicians earning money for the first time are often confused about exactly which expenses can be deducted or used to offset income made from earnings. Generally speaking, the federal income tax rules are pretty straightforward: ordinary and necessary business expenses can be deducted from your income. Before running off to write a huge check to your local music store, hiring a personal trainer, or picking up the tab each time you have dinner or drinks with your bandmates, step back to assess these expenses. Can they truly be written off? If so, how much of these expenses can be deducted?

The term *ordinary and necessary* is a very subjective one. Common business expenses, such as postage, office supplies, photocopying, studio rental, and professional fees (for accountants, lawyers, and the like) are rarely at issue. Expenses that are particular to the music business, however, are a different story.

You may think to yourself, "Hey, this is rock 'n' roll. Everybody is out of the ordinary, so all expenses in my pursuit of rock 'n' roll are okay." Use good judgment and be reasonable when deciding what to try to use as a deduction. What may be necessary and reasonable for a superstar artist (flying Lear jets to concerts, hiring a masseuse to accompany you on tour, supporting an entourage of employees, and leasing a sports car to maintain an image) will probably be considered neither ordinary nor necessary if you have not yet achieved that status.

The federal tax code puts some limits on your spending habits, too. Remember that there is a limit on what you can deduct dollar-for-dollar when it comes to gear. (In 1997, the limit was $17,500.) Two strategies for getting the most bang for your buck are spreading out your equipment purchases over a few taxable years and working with your salesperson to lease the gear. And the IRS allows only a certain percentage of travel and entertainment expenses to be deducted, not the entire amount. So although it has been fashionable to fight over the

## DETAILS, DETAILS

Here are a few suggestions to help you stay on top of your financial game and avoid some potential tax-related pitfalls.

**Keep good records.** Find a place to put all of your business-related receipts, and keep up your books regularly. A variety of inexpensive and easy-to-use small-business computer programs are available that can get your finances organized.

**Don't ignore your taxes.** Failure to report or pay taxes simply because you are a musician is not recognized as a valid reason by the government. When you are contacted by a tax board for any reason at all, always respond.

**Use professionals.** If you receive notification of an audit, nonpayment, or underpayment, consider hiring a pro to represent you. Just like hiring a top-flight studio musician or a good producer, using an accountant or tax lawyer with experience—particularly entertainment experience—will save you heartache, time, and money in the long run.

dinner tab in the past, think twice before picking up that $500 dinner and round of drinks to celebrate signing a record deal. Again, the key is how reasonable your use of the meal and entertainment deduction is.

## WORKING AS A GROUP

You are responsible for reporting income and paying taxes not only when you work as an individual, but also when you work as a group. Different tax issues arise, however, when working in a group. The tax forms required by the IRS and your state tax board will vary depending on the structure of your group (that is, a partnership, limited liability company, or corporation) and in which state it is organized. In some cases, the IRS merely requires that an informational return be filed; this type of return reports who received income from the group and how much they received. In other cases, the group itself is liable for the payment of taxes.

**Use good judgment and be reasonable when deciding what to try to use as a deduction.**

## AUDITING AND COLLECTIONS

The stereotype of the tax collector banging down the doors of delinquent taxpayers has persisted through the ages. At recent congressional hearings, IRS employees testified about outrageous and overreaching tactics that the bureau had used to collect back taxes. It is no wonder that many musicians hit the panic button when a letter arrives announcing an audit, nonpayment, or underpayment of taxes.

The purpose of an audit is to determine the accuracy of a tax return you have submitted. The audit may require that you meet with a representative of the taxation body and go over the questionable tax return to justify the figures you provided. Being organized and having backup materials on hand will help you in an audit.

If you are found to be liable to the IRS or to your state tax board for nonpayment or underpayment of taxes, you may have to pay not only the taxes, but also severe penalties and interest payments. Depending on the amount in question and how far back the tax debt goes, such drastic steps as putting liens on your wages, bank accounts, or royalties could be taken. You can either negotiate with the tax boards yourself or hire a professional (a tax lawyer or an accountant) to do so on your behalf. This may result in compromising to pay a lesser amount in taxes, paying off your tax liability in installments, or some combination of the two.

## WAIT, THERE'S MORE

Federal income taxes are the type that first come to mind when we think about taxes. As you plan your financial path, however, there are other important considerations. Some states have become aggressive about taxing the income that you earn as a musician within their boundaries, so you may have to stay on top of the reporting and withholding requirements of each state in which you make significant earnings. For example, if you travel to a number of states on a major grossing tour, you could be liable to each state based on the money you earned in that state.

In addition to income taxes, you may have to pay business taxes to county and city governments for the privilege of doing business within their boundaries. If you are selling your products, such as CDs, T-shirts, and other merchandise independently, those sales are probably subject to the payment of local and state sales taxes.

Above all else, become familiar with the federal and state tax codes as they apply to your music business. Many books are available that explain the tax structure. Arts organizations and accountants looking for business sometimes give tax seminars for artists. By being organized and aware, you will hopefully spend more of your time on your music and less of it looking over your shoulder.

---

**Michael A. Aczon** *is experiencing the thrill of his 12-year-old daughter's discovery of how cool learning to play the guitar can be.*


EDIROL Introduces
USB Audio & MIDI Products
for Great Music Production
NEW TO NORTH AMERICA!
from Roland®
USB
UNIVERSAL SERIAL BUS
One Cable, Pure Sound!
USB is a next-generation interface for connecting peripheral devices to a personal computer. Roland's USB audio & MIDI products allow for clean, high speed transfer of data.
USB Audio & MIDI Processing Unit
AUDIO Canvas
UA-100
• 20 Bit External Digital Audio Converter
• Clean Digital Audio with No CPU "Noise"
Powered USB Speakers
MA-150U
• 15 Watts
• Plug & Play
• 20 Bit External Digital Audio Converter
64 Channel USB MIDI Interface
Super MPU64
• Connect Four MIDI Devices to Your Computer
Large Selection of Desk Top Media Production Systems, Software & MIDI Files!
AUDIO Canvas
VIDEO Canvas
SOUND Canvas
Your Complete Source for DTMP
Call for our complete Desk Top Media Production catalog
Contact us for our FREE 76-page, full color catalog.
Distributed by
EDIROL
Member of the Roland Group
1-800-380-2580
www.edirol.com
Dealer inquiries welcome
SEE US AT NAMM BOOTH #1521
circle #571 on reader service card


# At Home in Your Range

***Shift your voice into high gear, but don't go beyond its limits.***

By Joanna Cazden

Every voice teacher is asked, "Can you show me how to hit really *high* notes?" And the answer is, "Maybe, maybe not." The pleas continue: "My rock band is getting some local buzz, but I need to sound like Steven Tyler of Aerosmith, or else we'll never get a record deal."

One singer I know was told that a hit record requires a high, edgy male vocal because that type of sound cuts through best on the average car radio. Pushing the upper limits of the voice seems to be a male preoccupation; female performers these days are often expected to sound low and husky. These goals bring to mind Procrustes, a nasty character in ancient Greek mythology who required that all his guests fit in the same bed. Short visitors had their limbs roped to the bedposts and stretched; those who were too tall got their feet chopped off. This is not a pretty story, but it's a good analogy for the demands imposed by some musical fashions.

**Sarah McLachlan's vocal appeal includes the way she uses the marked contrast between her registers to express different moods.**

The obvious message is that straining your voice in either direction might win an immediate gig or contract but can hurt your voice in the long run. A cello can't be played effectively in a piccolo's range any more than a Sumo wrestler can dance ballet *en pointe.* Each voice has its own limitations, and to make your best music, you must respect those limits. The first step is to identify your natural vocal range.

## MUSCULAR ACOUSTICS

Let's begin with an overview of some anatomy. The *vocal folds* (or "vocal cords") are small, semi-elastic muscles. When you sing a scale or melody, your vocal folds are stretched longer and thinner for higher pitches, and they relax and thicken for lower pitches.

When your vocal folds are inflamed and swollen (when you have a bad cold,

steps or a perfect fourth). You can realistically expect to go two to three times as far *up* from the reference pitch before you reach the first "break point" of your range. You might be able to sing an octave or so above that break point, but it will be in a different register (to be discussed in a moment).

Keep in mind that your basic range will fluctuate slightly from morning to evening and from one day to the next, according to your health, mood, and level of stress. For the most reliable measurement, repeat this process at different times of day and over several days, then use the average result.

## REGISTRATION, PLEASE

As you experiment with singing in different pitch ranges, your voice might sound weak or strained on a particular note, but then something shifts inside your throat and your voice stabilizes a step or two further up (or down) with a different tone quality. That is because you're moving from one *register* into another. You've probably heard registers referred to as *chest voice* or *belting*, in contrast to *head voice* or *falsetto*. Voice scientists and vocal instructors still disagree on the precise terminology for these register changes, nor is their production fully understood.

However, at the level of sensation and "throat feel," the register shift feels similar to operating the manual transmission of a car: As the driver accelerates in one gear, the engine revs faster. This is roughly analogous to a normal increase in effort and vocal-fold tension as someone sings up a scale. Then, at a convenient point, the driver shifts into the next gear, changing the ratio of power and speed, and the motor runs at a more relaxed rate. Similarly, a shift from chest voice to falsetto takes pressure off the vocal folds and requires subtle adjustments in breath support.

Each car's transmission shifts fluidly at a slightly different speed, and the driver gets the feel of it with practice. Just as a driver has a 10 to 15 mph range in which it's safe to shift gears, switching vocal registers can be accomplished anywhere within a small range of pitches, depending on the lyrical and musical context. The breaks in the lines in **Figure 2** illustrate where register shifts are generally expected in different vocal ranges.

# ACCURATE MONITORING IS NOW A RIGHT, NOT A PRIVILEGE.


Accurate monitoring used to be expensive. Not any more. Available in both active and passive versions, the new HHB Circle 5 incorporates a number of unique breakthroughs in loudspeaker technology to create a compact, high performance studio monitor that's ideal for use in a wide range of professional applications. An investment of $250,000 in research and development has produced a loudspeaker of exceptional clarity, with a sound that doesn't fatigue the listener, even after a long session.

So if you're looking for a great sounding studio monitor, listen to the Circle 5 at your nearest HHB dealer and prepare to be impressed. Then ask the price and prepare to be amazed.

HHB CIRCLE 5

- Varied cone thickness minimises low frequency distortion
- Low Q filters deliver an untiring sound during long listening sessions
- Detailed and accurate on and off axis sound
- 120W LF / 70W HF integral amplifier pack (active version)
- Delivers 'large monitor' performance from a compact loudspeaker
- Individually tested and matched ferro-fluid cooled soft dome tweeters
- Magnetically shielded for use near computer and video monitors
- Solid State Polyswitch® tweeter overload protection

British sound at its best

HHB Communications USA LLC · 626 Santa Monica Boulevard, Suite 110, Santa Monica, CA 90401, USA
Tel: 310 319 1111 · Fax: 310 319 1311 · E-Mail: sales@hhbusa.com

HHB Communications Canada Ltd · 260 King Street East, Toronto, Ontario M5A 4L5, Canada
Tel: 416 867 9000 · Fax: 416 867 1080 · E-Mail: sales@hhbcanada.com

HHB Communications Ltd · 73-75 Scrubs Lane, London NW10 6QU, UK
Tel: 0181 962 5000 · Fax: 0181 962 5050 · E-Mail: sales@hhb.co.uk

http://www.hhb.co.uk

HHB

circle #576 on reader service card

Classical, jazz, gospel, and pop singing generally require a consistent loudness and smooth transition between registers. (This is one of the many strengths of Whitney Houston's voice, for example.) Other genres—such as country, bluegrass, and some folk styles—make use of register breaks that are more abrupt. (Think of cowboy songs, Swiss yodeling, and early Joni Mitchell recordings.) One of the appealing aspects of Sarah McLachlan's vocal style is the marked contrast between her registers, a contrast she uses to express conflicting emotions.

Your own break points will fluctuate slightly: they will be lower in the morning and when you are relaxed or ill, and they will be higher when your adrenaline is pumped. The best way to deal with these shifts is to become familiar with them, avoid forcing your voice roughly in the transitional areas, and develop each register fully on a foundation of good overall technique. The transitions will become easier, and you will be able to use a greater variety of tones for your own creative purposes.



## MUSICAL SOLUTIONS

Now that you know which notes you can realistically use and where your voice is apt to change register, go through your repertoire and transpose each song to best fit your voice. A digital keyboard or sequencer can make this process a lot easier than it used to be. If you're in a band, you should work as a team to find the right key for each song, instead of merely retaining the song's original key. This might seem like a lot of extra work at first, but it will be worth it in the long run. Eventually, fitting the music to your voice will be as routine as an athlete selecting the right shoes. And experimenting with different keys and instrumental voicings can make your music more appealing.

If you haven't had much training, work with a private teacher to get your voice in better shape and thus increase your usable range a bit. And if you already have reasonably good vocal technique, practice scales every day at medium loudness, with proper posture and breath support; doing so can help you to gradually add a whole step or two at the top and bottom of your range.

Classical singers and MTV divas are expected to use a two- to three-octave range, gig after gig, with both power and control. However, the average pop melody rarely requires more than an octave or tenth (not counting key modulations and vocal ornaments). For example, Tracy Chapman sings within a relatively small range, but most listeners either don't notice or don't care because of her rich timbre, bluesy inflection, and deeply honest presentation.

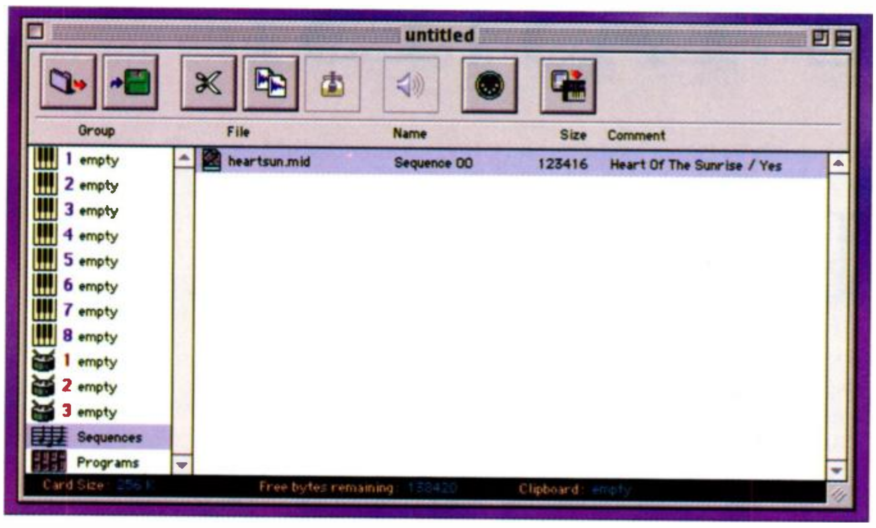
Using your true voice within a healthy range will ultimately sound more powerful and expressive than shaping your vocals to the arbitrary demands of the marketplace. Fiona Apple doesn't try to sound like Stevie Nicks, and Luther Vandross is no Johnny Cash. Your voice is already a custom design, so keep it healthy, capitalize on your strengths, and let your unique music shine through.

---

**Joanna Cazden** *is a singer-songwriter and vocal coach in Southern California. She can be reached at jcazden@att.net.*

OPERATION HELP






**Alesis's *SoundBridge* enables you to fly your audio tracks along with your sequences onto the QCard flash memory card.**

starts from: Left, Right, or Alternating Left/Right. You can also set the Sweeping Mode (Continuous or One Cycle Only) and Trigger Source (Left or Right input channel).

Here are a few applications in which the Triggered Panner really stands out:

1. Set the Retrigger Point to Alternating, Doppler to 0, and the Sweeping Mode to One Cycle Only. Send a repeating vocal sample to the Q20, and set the Speed so that the pan lasts just as long as the sample. Now every time that sample plays, it will pan from left to right and then from right to left.

2. If you're panning a signal with lots of peaks (like drums or rhythm guitar) that retrigger the pan too early, try setting the Trigger Source to a Footswitch with nothing plugged into the jack.

3. Use the Doppler parameter to add a subtle flanging effect to panning drums.

4. The Distance parameter causes the signal to fade out when panned to the extreme right or left. This is almost like a rotary speaker effect at higher speeds. Of course, it isn't as convincing as the Q20's Rotary Speaker Simulator, but it's much more controllable.

5. Try using a very slow speed that doesn't retrigger, then pan the Q20 return channels so that they hardly pan at all. This can add life to a flat acoustic guitar without sounding like an obvious effect.

Best of all, the Triggered Panner uses only 10 percent of the DSP, so you can add other effects (such as a triggered panner into a reverb or flange block) or use the other input as a separate reverb send. Have fun with your next mix.—*Courtesy Jeff Laity, Alesis Corporation*

## ALESIS QS-SERIES SYNTH

With the *SoundBridge* software that is included with all Alesis QS synths, you can store SMF files of any length, as well as any AIFF or WAV file. Your only limitation is the size of your card. This means that if you're using a computer-based sequencer that allows you to integrate audio tracks, such as MOTU *Digital Performer* or Cakewalk *Pro Audio*, you can fly your tracks along with your sequences onto the Qcard (flash memory card). You can then run applications such as automating background vocals for live performance while processing them with the QS effects processor.

You can use a card that holds up to 8 MB in each slot on your synth. So if you're using two 8 MB cards on a QS8, for example, you can store around three minutes of mono 44.1 kHz samples and still leave room on the card for a few sequences.

Here's a cool application for the card slot on the back of your QS synth. This is pretty easy to do, even for the novice programmer:

1. Create a *SoundBridge* project with the samples you want to use. Burn them to a card, making sure to save the project.

2. Initialize the synth by holding down buttons 0 and 3 during power-up. Then go into the Voice menu, and assign the samples to a program. Customize the program and effects parameters to fit your needs.

3. Trigger the samples as part of your computer-based sequence. Just play the notes that you have assigned to trigger the samples in the same manner that you would record any MIDI part.

4. Save the sequence as a Standard


**You can use the Roland VS-840's Cue knobs to set pan levels for all eight mixer channels.**

Now Hiring

# Do you want a great job?

Are you crazy about consoles?

Do you swoon over synths?

Are you mad about multitracks?

Are you delighted with delays?

Now's you chance to do more than just land a job. Here at Sweetwater, you could find a fulfilling career working with the latest music technology and pro-audio equipment along with an excellent team of knowledgable, friendly people in a beautiful city with a low cost of living!

Thanks to our exceptional growth, we can offer you an exceptional career. Want to make the best move for your career and your family? If you love music technology and know your stuff, this is your move! Why not call us right now and make an appointment? Ask for the Career Center.

Sales · Recording Studios
Service/Technical Support
Contracting/Installations
Desktop Publishing
Web Design

**Why not take the next step in building your career? Give us a call today!**

# (800) 222-4700

**Meet us in L.A.**
*January 28-31, 1999*
At NAMM booth# 1827 or call for a personal appointment!

Music Technology
Direct & the Best
Value Guaranteed!

All applications taken in strictest confidence.

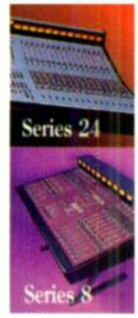
Sweetwater SOUND INC.

5335 Bass Road, Fort Wayne, IN 46808
(219) 432-8176 • FAX (219) 432-1758
www.sweetwater.com • sales@sweetwater.com

**SEE US AT NAMM BOOTH #1827**

**circle #649 on reader service card**

BEQ Mixers
*Analog perfection inside a luxurious retro flavored mixing desk.*




# ORAM

## How do you get that BIG Pro Sound?

If your quest is that "Big" sound, we recommend the fine line of signal processors and consoles from England's John Oram. Oram set world-wide sonic standards designing legendary amplifiers for VOX and consoles for Trident where he earned the title "Father of the British EQ." How does ORAM achieve that great sound and feel? He combines 90's design techniques with good old, brute force pro audio. That means overkill at every possible juncture. And the ergonomics are exceptional. No flimsy plastic knobs or budget faders. You get lush, top quality controls. These products feel and look as big and luxurious as they sound.


Until you hear for yourself and experience the smooth, lush quality of the controls, you'll never quite understand what makes these devices so extraordinary. Wouldn't you like a desk or module that not only makes a true sonic difference in your recordings, but also provides an extraordinary experience in its use? Aren't you on a quest for that perfect blend of power and luxury? Try Oramsonics yourself and bring that "big" sound to your very own studio.

**HD-EQ II Equalizer • Sonicomp 2 Compressor**
**MWS II Microphone Workstation/Equalizer**
**Octamix mixer • Octasonic mic preamp**
**BEQ mixing consoles: 8, 24, 48-bus configurations**

Call today and own the excellence of Oramsonics!

Sweetwater SOUND INC.

*ORAM products are distributed exclusively in North America by SWEETWATER SOUND!*

**MUSIC TECHNOLOGY DIRECT AND THE BEST VALUE GUARANTEED!**

# (800) 222-4700

5335 Bass Road • Fort Wayne, IN 46808 • (219) 432-8176 • FAX (219) 432-1758 • www.sweetwater.com • sales@sweetwater.com

**SEE US AT NAMM BOOTH #1827**

circle #650 on reader service card

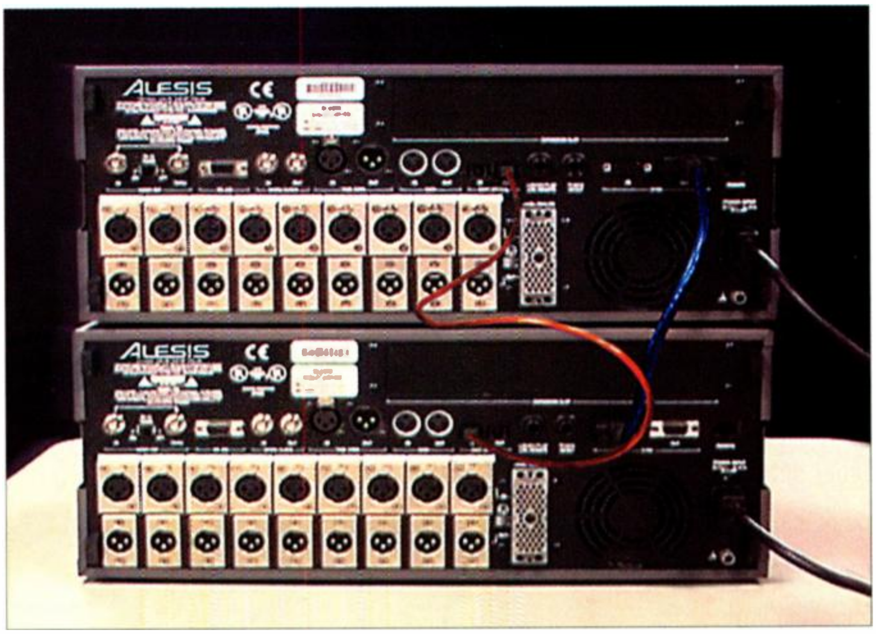
MIDI File. Your sequence can now be seen by *SoundBridge*.

5. Reopen the *SoundBridge* project that you began compiling earlier. Add the sequence to the project and reburn the card.

It's that simple. Now, when you play your sequence from your synth, the samples will automatically be triggered at the correct time. So next time you hit the road with your QS keyboard or module, just take your flash card along and leave your computer and sampler at the studio.—*Courtesy David Bryce, Alesis Corporation*

## CREATING TIME-REFERENCED ADAT CLONE TAPES

The ability to make clone copies of an ADAT master tape is a well-known feature of every ADAT digital multitrack recorder. However, when a BRC (or a CADI in the case of the M20) is unavailable, guaranteeing that the sample-accurate clone has maintained phase accuracy with the original (that is, keeping the exact 32-bit absolute time reference for every sample) requires that you perform a simple additional step before the transfer is started.




**In order to achieve a phase-accurate digital transfer, the ADAT system must be configured in a master/slave relationship.**

A typical setup might use two ADAT recorders. To achieve a phase-accurate digital transfer, the ADAT system must be configured in a master/slave relationship, with the two ADATs synchronized via the built-in 9-pin sync interface. Do this by


**By playing a single note into the Boss HR-2 Harmonist, you can get three-voice intelligent harmonies that are correct to key and scale.**

connecting an Alesis-approved sync cable from the sync out of the master ADAT (ID1) to the sync in of the slave ADAT (ID2). Connect a fiber-optic cable from the digital out of the master ADAT to the digital in of the slave ADAT. Insert the original master tape in the master ADAT and the backup tape in the slave ADAT. With this typical system configuration, the slave ADAT will receive sample clock and Alesis proprietary 32-bit time code from the master's sync out and up to eight tracks of sample-accurate digital audio from the master's digital out.

The next step is the most important and is what guarantees a time-aligned (that is, phase-accurate or time-referenced) transfer. Enable the digital input on both the master (source) and the slave (destination) ADATs. This procedure differs according to the model of ADAT used. On the original ADAT, the master machine must be set up to use its internal clock when in Digital In mode. To do this, press and hold Set Locate, then press Digital Input. The seven-segment counter will briefly indicate "INT," meaning that the master will use its internal crystal even though digital input is enabled. Next, enable digital input on the slave. On the ADAT-XT, XT20, and LX20, press the Digital Input button on the master and slave. Make sure that the clock is set to INT in the display of the XT, XT20, or LX20 master.

Setting up the M20 is a bit more involved. On the master and slave M20, enter Input Select mode, and assign channel pairs for digital input using the input enable buttons. The change to digital input is indicated by the lit D indicator next to each channel's number designation in the meter display. Verify that the clock source on the master is set to INT, or that sample clock is derived from a stable external source (for example, word clock or video). The digital source on the slave M20 should be set to ADAT Optical.

By enabling the digital input and keeping the clock referenced to the internal crystal on the master ADAT, you have aligned the digital output of the source ADAT to the destination ADAT (compensating for inherent system latency). Thus, you've achieved a perfect sample-accurate and time-referenced transfer for your clone backup tape.—*Courtesy Don Hannah, Alesis Corporation*

## ROLAND VS-840

Once you are comfortable with basic operations, you'll find that the VS-840 has tons of features to help you make better music. One of these "hidden" features is the ability to use the Track Cue knobs to set pans, effects, aux send levels, and even EQ settings.

Here's how to use the Track Cue knobs as eight pan knobs:

Hold Shift and press the Pan button that is located above the Channel Mixer section. Now, while continuing to hold Shift, you can use the Track Cue knobs to set pan levels for all eight mixer channels.

To use this same feature for setting other mixer levels or EQ, hold Shift and press either the Effects or Aux Send levels, or the EQ button over the channel faders. The key is to remember to keep pressing the Shift button while adjusting the levels with the Track Cue knobs.—*Courtesy Tom Stephenson, Roland Corporation* www.digitech.com

DigiTech
A TRADITION OF INNOVATION

# WE'VE GOT YOUR BASSES COVERED ... AND SOPRANOS TOO

## VOCALIST HARMONY SERIES

### VOCALIST WORKSTATION EX

The ideal tool for musicians, singer/songwriters and MIDI/project studios, the Vocalist Workstation EX lets you create up to 4 harmony voices that have fully editable MIDI parameters, reverb and effects. The Workstation's built-in mixer can be can be controlled easily from the front panel faders and the desk-top format makes editing a snap.

### VOCALIST PERFORMER

Affordable and easy to use, Performer is a natural sounding 2 voice vocal harmony processor with reverb. Performer does not require MIDI control and comes with a handy micstand mount for live performance.

### VOCALIST ACCESS

New to the Vocalist family, Access is a great solution for musicians, producers and engineers who rely on MIDI for control. Access gives you the same great 4 harmony voices, reverb and effects neatly tucked in an attractive 1U package.

### STUDIO VOCALIST EX

With its 4 voices of harmony, The Studio Vocalist EX is the flag ship of the line with an array of harmony, effects and editing features mounted in a sturdy 2U box. Features such as the 5 independent XLR outputs, 48V phantom power, genderbender (formant shifting) and the digital I/O option make the Studio Vocalist EX the best choice for the discerning professional.

*Your voice is uniquely yours. It's your instrument, your signature. Your style may be loud and proud, bluesy or soft and sultry but no one else in the world sounds quite like you.*

*DigiTech now offers you a dazzling array of vocal harmony and effects processors tailored to suit your individual needs. From the easy to operate live performance models to the more in depth studio versions with editable parameters, pitch correction and digital I/O options, there is a Vocalist for you.*

*Whether you're a performer, a singer/songwriter with a MIDI studio or you operate a professional recording facility, we've got what you need. Each member of the family has the same outstanding sound quality with separate features for your specific applications.*

*But don't just go by what we're saying. Let your ears be the judge. Check out the Vocalist Harmony Series at your DigiTech dealer today.*

## ...DEFINE YOUR VOICE

©98 Digitech 8750 South Sandy Parkway, Sandy Utah, USA 84070 (801) 566-8800 Fax (801) 566-7005 www.digitech.com

circle #585 on reader service card

A Harman International Company

● OPERATION HELP

## USING GUITAR PEDALS WITH OTHER INSTRUMENTS

They're easy to use, they're inexpensive (okay, some are downright cheap), they're built like tanks, and they can make sounds that your keyboard's onboard effects processor can't. So why aren't you using them?

We're talking about guitar pedals, those wicked little boxes on the floor that guitarists have used for years to radically alter the sounds of electric guitars. Pedals such as distortions and overdrives, auto-wahs, pitch shifters, and others are useful for things besides guitars. They can create cool new sounds quickly and easily, as many adventurous keyboardists and other musicians are discovering.

Consider just one example: Boss makes a compact pedal called the HR-2 Harmonist. By playing a single note into the HR-2, you can get three-voice intelligent harmonies that are correct to key and scale. Because this is a pedal, you can turn it on and off by stomping on it, rather than having to program it into a patch on your workstation or studio rack. You can even sing into it with the same results: you'll hear your original voice and two others, in perfect harmony.

Another interesting application is distortion. Guitarists use distortion effects to create thicker, bigger sounds with more sustain. What if you used a guitarlike sound from a keyboard through a distortion pedal? One interesting result would be a more convincing guitar sound, especially for leads. Interestingly, some distortion and overdrive pedals work much better than others for keyboards, due to differences in the sound characteristics of the pedals. Among others, the venerable Boss SD-1 Super Overdrive and the Boss MT-2 Metal Zone are particularly effective with keyboards, especially the MT-2 (due to its parametric EQ).

These are but a few ideas; many other stomp-box-type effects such as flangers, delays, and EQs can be used with great success on instruments other than guitars. Try a Boss BF-2 Flanger on your next drum loop or background vocal for something different. There are dozens of other pedal effects to try out. You don't have to be a guitar player to take advantage of their sonic potentials. Break some rules, get some pedals, and stomp on 'em!—*Courtesy Peter Swiadon, Roland Corporation*

# REVIEWS



## YAMAHA

### DSP FACTORY (WIN)

***Digital mixer meets budget digital audio workstation.***

By Jeff Casey

When the Yamaha 02R hit the market some years back, what impressed most people about the product was the sheer amount of DSP power that had been packed into a relatively small console. Until that point, such processing prowess could be found only in larger, more expensive digital mixers.

Well, Yamaha is at it again, downsizing and price busting, and the release of its long-anticipated—and very reasonably priced—DSP Factory marks the company's first foray into the world of computer-based recording. The DSP Factory essentially puts the processing power of an 02R console onto your PC. But that's not all. You also get a 16-channel, 32-bit hard-disk recording system (useful when 32-bit software becomes available) with expandable multichannel analog and digital I/O—all for less than $1,300. Several third-party manufacturers already produce supporting software, and with a Macintosh version due out soon, the DSP Factory looks to be a computer-based production system to be reckoned with.

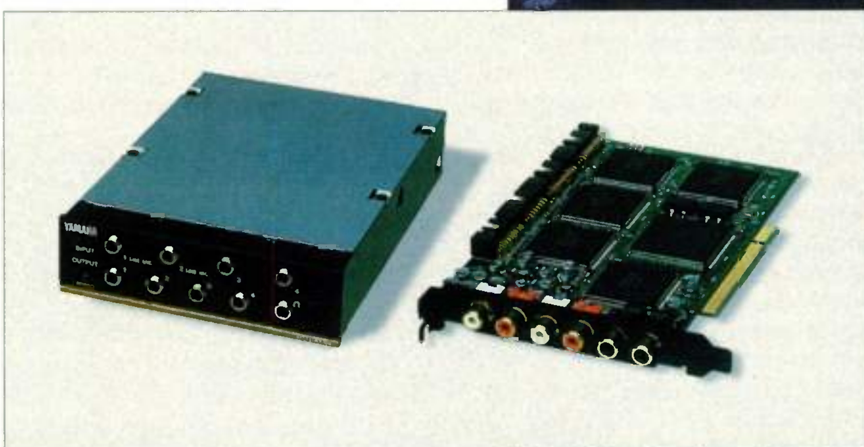



**FIG. 1: The DS2416 PCI Audio Card (right) is the heart of the DSP Factory. Besides housing the system's five DSP engines, the DS2416 offers analog and digital I/O on RCA connectors. The AX44 Audio Expansion Unit (left) provides four additional channels of analog I/O on ¼-inch TRS jacks, as well as a ¼-inch headphone jack. A/D conversion is done at 20-bit resolution, and D/A conversion is at 18-bit.**

FIG. 2: The AX44 Audio Expansion Unit is mounted in a computer drive bay. Up to two AX44s can be used for each DS2416 card.

## AN 02R ON A CARD

The DSP Factory is centered around the DS2416 PCI Audio Card (see **Fig. 1**), which houses the inner workings of both a Yamaha 02R and a 16-channel hard-disk recorder. All internal processing is handled at 32-bit resolution (like the company's 03D mixing console), and five discrete DSP engines power the system. Because all recording and mixing functions are performed by the DS2416, your computer is free to handle other system chores, such as managing files or running associated software.

The DS2416 delivers a professional matrix (24 input channels, 8 bus outputs, and 6 aux sends); 26 individual dynamics processors (24 channels and the stereo bus); 4 bands of parametric EQ on each channel and on the stereo bus; 20 channel delays; and 2 multieffects processors. Dynamic mixdown automation is also available. There are level meters for every input and output, and gain-reduction meters for each dynamics processor. (These elements are displayed through the controlling software, which I'll discuss shortly.)

On the recording side, the DS2416 supports 16-channel simultaneous playback and 8-channel simultaneous record.

Interestingly, the mixing and recording sections of the DS2416 are totally independent of each other. For example, you can record directly to disk from an external source, bypassing the mixer completely, or you can use only the mixer to process signals from an external multitrack. This versatility is an especially nice touch, considering the substantial mixing power that the DSP Factory offers.

Each DS2416 card occupies a single 5-volt PCI expansion slot in a PC tower. (It cannot be installed in a 3.3-volt slot.) Two cards may be cascaded for a total of 48 channels of mixing and 32 channels of hard-disk recording, or you can digitally interface the system with another sound card (for example, a Creative Labs Sound Blaster or Korg 1212 I/O). The DSP Factory has no general minimum system requirements for the PC's processor type, memory, and hard-disk capacity—they depend on the controlling software you use.

The DS2416 provides two channels of 20-bit analog I/O (on RCA connectors); stereo digital I/O (coaxial) is switchable between 20- and 24-bit operation. These connections appear on your computer's rear panel—a bit of a hassle—so Yamaha has provided modular expansion units as a more accessible alternative.

## INTERNAL EXPANSION

The other half of the DSP Factory is the AX44 Audio Expansion Unit, which provides four additional channels of 20-bit analog I/O on a front-mounting drive bay module (see **Fig. 2**). Up to two AX44s can be connected to each PCI card, so if you have two DS2416s (and four open drive bays), you can gain an extra 16 channels of analog I/O.

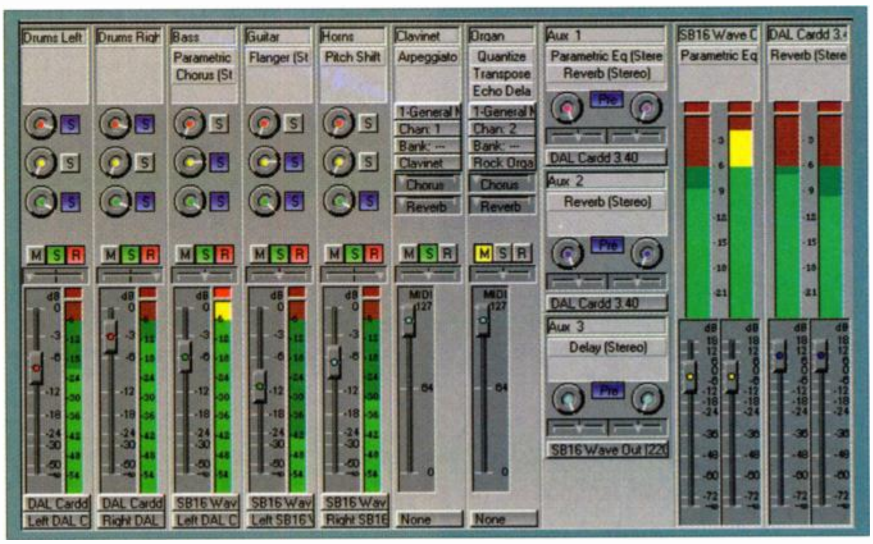
Audio connections are made through unbalanced ¼-inch TRS connectors. The AX44 also has a ¼-inch headphone jack with an associated volume knob. You can alternate inputs 1 and 2 between mic- and line-level operation by toggling switches located next to their jacks. The dynamic range of these connections is an impressive 100 dB. The AX44 connects to the DS2416 with a supplied 20-pin ribbon cable. You'll also need to connect the module to your computer's power supply with a standard 4-pin DC drive bay cable.

In addition to the AX44, Yamaha has several other I/O modules currently in development. The first one planned for release is the AX16-AT Audio Expansion Card, which will provide 16 channels of ADAT Lightpipe I/O. The AX16-AT is expected to arrive in stores in April.

## WELL, NOT EXACTLY

It's important to note that although the DS2416 is the essential guts of the 02R, the DSP Factory is not a replacement for Yamaha's flagship digital mixer. For one thing, an 02R has far more I/O ports. Sure, the DSP Factory includes 02R internal buses, but you don't get a lot of I/O, and the DSP Factory's expansion ability is limited by your computer's drive bays.

Keep in mind that a stock 02R gives you 24 analog channel inputs (with separate mic and line inputs on the first eight channels), insert jacks on the first




Cakewalk Music Software's *Pro Audio* 8 supports the DSP Factory's I/O and signal-processing capabilities but currently does not support the DS2416's mixing functions.

eight channels, and six auxiliary send jacks. You also get two stereo aux returns, two pairs of 2-track inputs, separate stereo control-room and monitor output pairs, two pairs of L/R mix outputs, and a stereo headphone jack. That assortment doesn't even include the various digital I/O ports, and it doesn't account for what you can add with the 02R's four expansion slots. Without all these ports, you can't take full advantage of the busing architecture.

Another very important point to consider is that an 02R has a hardware control surface. With the DSP Factory, you have to either mix with a mouse, use some sort of MIDI controller box, or buy a Yamaha digital mixer (which can cascade with the DSP Factory).

## A SHOW OF SUPPORT

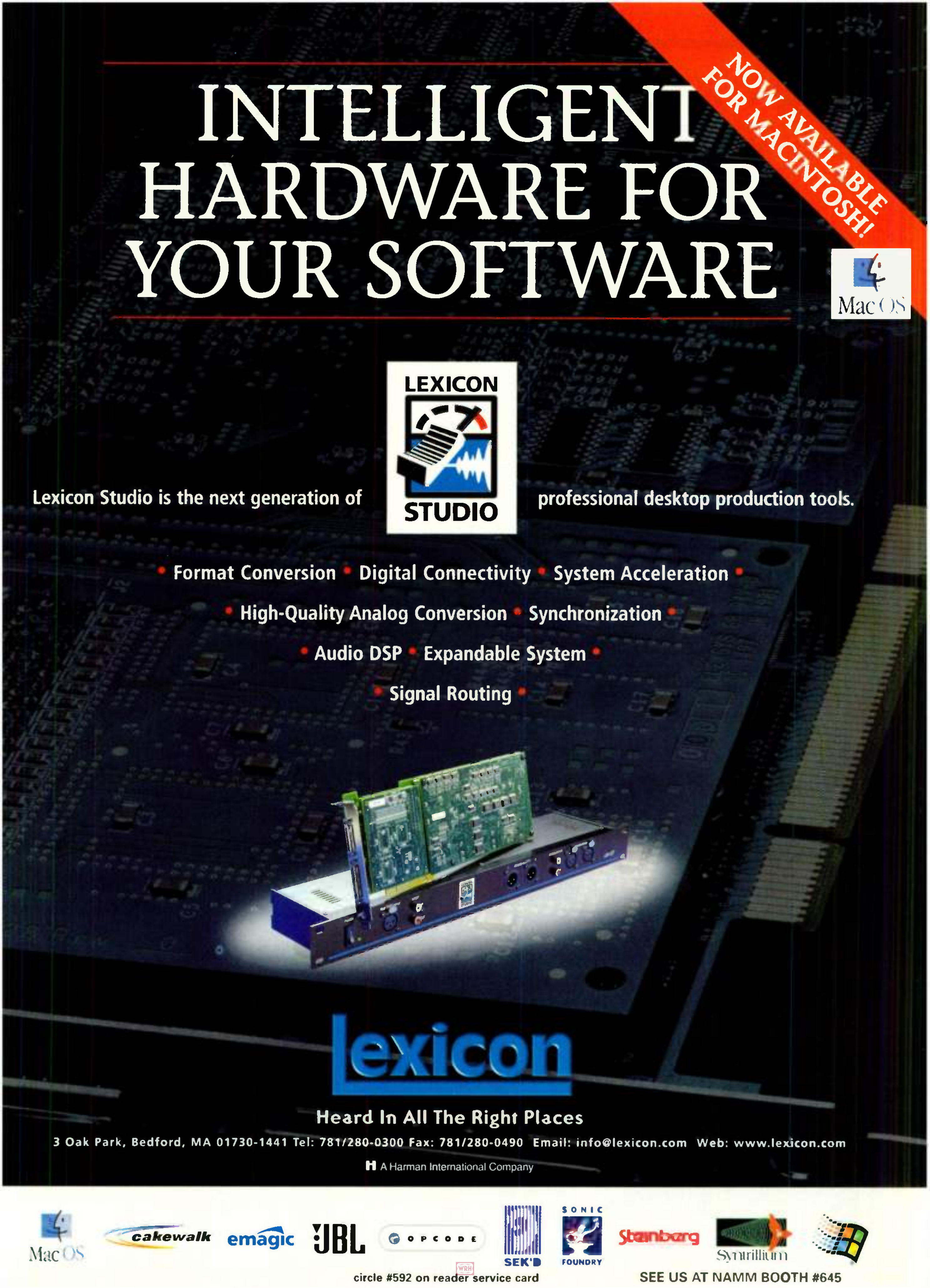
Unlike many comprehensive computer-based recording systems, the DSP Factory ships with no controlling software, which may or may not be a bad thing. For those of us who already own and use digital audio recording and MIDI-sequencing software, the omission of a "starter program" is no big deal; quite frankly, I can't remember the last time I used *Pro Tools* software with a Pro Tools system.

For those folks just getting their feet wet in this often confusing ocean of computer-based recording though, being unable to take full, immediate advantage of all DSP Factory functions (without spending a few hundred more dollars) might be a nuisance. I, for one, am pleased with the route that Yamaha has taken, because the inclusion of software would inevitably have jacked up the system's cost.

The DSP Factory can use any software that supports Windows Multimedia Extensions (including Media Player) for recording and playback. To take advantage of the system's mixing and DSP functions, however, you'll need to have software that's specifically designed for use with the DS2416.


**FIG. 3: Steinberg's *Cubase VST/24* is one of several programs that offer support for the DSP Factory. In the Upper display of the DSP Factory Input Console, you can view channel EQ, dynamics controls, aux send levels, or bus send levels.**

Ten companies have developed such software, including Steinberg (*Cubase VST/24*), Cakewalk Music Software (*Pro Audio* 8), Emagic (*Logic Audio Platinum* and *Gold*), Canam Computers (*Studio Pro*), C-Mexx (*C-Console for DSP Factory*), and Minnetonka Software (*MXTrax*).


INTELLIGENT HARDWARE FOR YOUR SOFTWARE
NOW AVAILABLE FOR MACINTOSH!
Mac OS
LEXICON STUDIO
Lexicon Studio is the next generation of professional desktop production tools.
• Format Conversion • Digital Connectivity • System Acceleration •
• High-Quality Analog Conversion • Synchronization •
• Audio DSP • Expandable System •
• Signal Routing •
lexicon
Heard In All The Right Places
3 Oak Park, Bedford, MA 01730-1441 Tel: 781/280-0300 Fax: 781/280-0490 Email: info@lexicon.com Web: www.lexicon.com
A Harman International Company
Mac OS
cakewalk
emagic
JBL
OPCODE
SEK'D
SONIC FOUNDRY
Steinberg
Syntrillium
circle #592 on reader service card
SEE US AT NAMM BOOTH #645

Some of the programs offer complete multitrack recording and MIDI sequencing environments, incorporating the DSP Factory's features into existing software, while others have been designed specifically to provide a control surface for the DS2416. My advice is to check the Web sites of the companies whose products you are interested in to find out how much support is offered and whether the software or upgrades have actually begun to ship.

Software that doesn't support all the features of the DSP Factory can still use a basic (albeit somewhat limited) feature set. Realize that in these instances, however, input and output assignments are often fixed and cannot be reassigned. If this dilemma applies to you, I strongly suggest that you refer to the DSP Factory manual (which provides a signal flow chart); otherwise, you'll be flying blind. You'll also have to rely on the Windows 95 Volume Control as your stereo master fader.

I tested the DSP Factory with Steinberg's *Cubase VST/24*, simply because I work with the program a lot and felt comfortable making informed judgments about the hardware using it as a controller. *Cubase* fully supports all the DSP Factory's features.

## THE TEST DRIVE

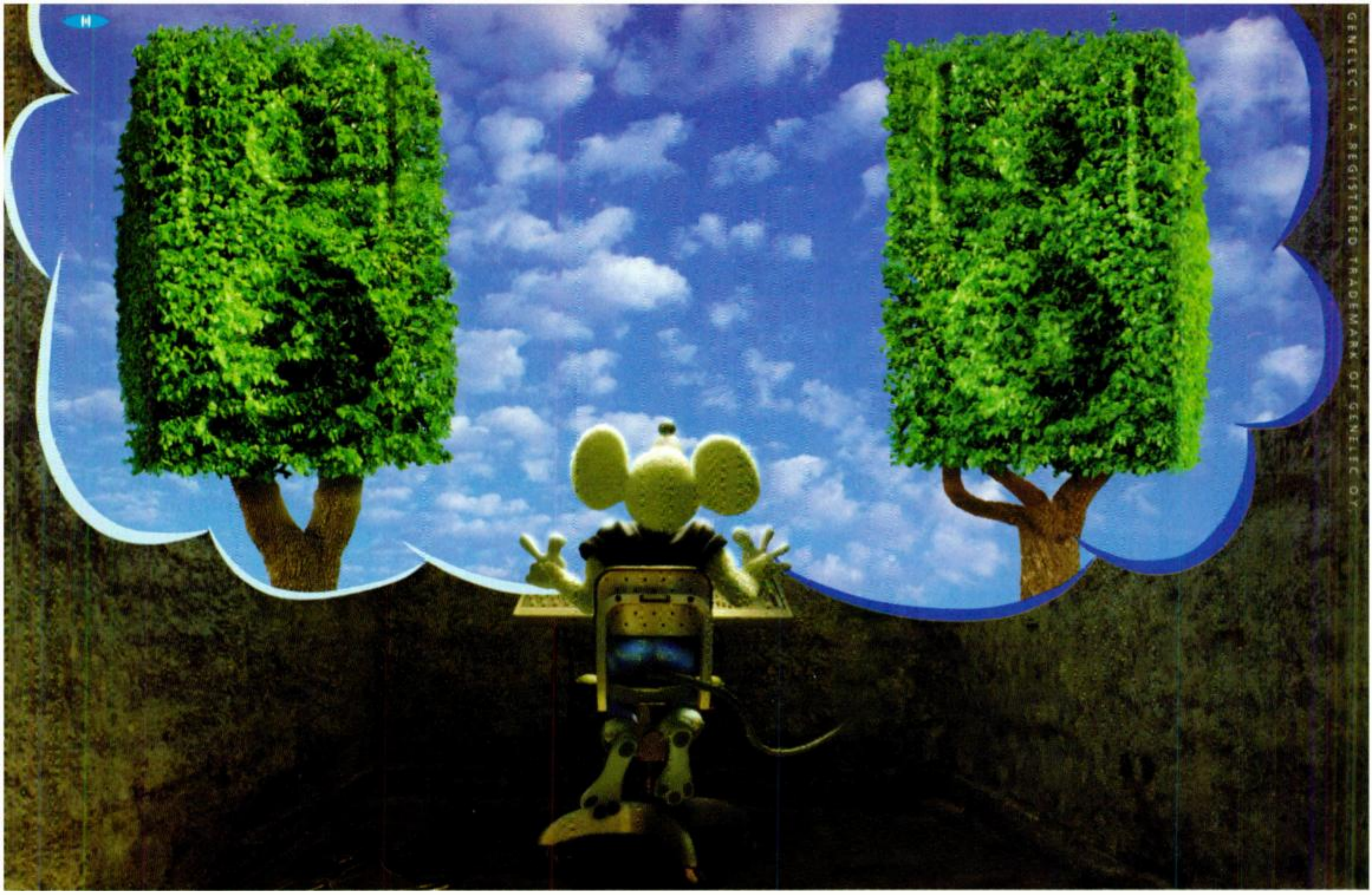
For this review, the DSP Factory arrived already installed on a Pentium II PC. However, I took the liberty of removing the DS2416 card and the two AX44 modules, as well as deinstalling the drivers, to see exactly how easy the system is to set up. I spent no more than half an hour before I was able reboot the computer and start getting down to business.

One bit of software that Yamaha does provide is a handy floppy-disk test program that ensures all hardware is connected properly; it worked to my advantage, considering that I had forgotten to connect one AX44 jumper to the DS2416. The program can also generate a handy 1 kHz test tone from

### DS2416 Specifications

| | |
|---|---|
| **GENERAL** | |
| Platform | Windows 95 |
| Number of Tracks per DS2416 PCI Card | 16 |
| Sampling Rate | 44.1 kHz, 48 kHz |
| **AUDIO** | |
| Vari-Pitch | 41.45–50.88 kHz |
| Frequency Response | 20 Hz–20 kHz |
| THD | <0.02% (20 Hz–20 kHz) |
| Dynamic Range | 93 dB |
| **MIXING SECTION** | |
| Mixing Channels | 24 plus stereo bus |
| Buses | 8 |
| Aux Sends | 6 |
| Channel Functions | level, mute, solo, pan, EQ, dynamics, delay (channels 1–20) |
| Stereo Bus Functions | level, mute, EQ, dynamics |
| Metering | -72 to 0 dB (32 steps) |
| **DSP EFFECTS** | |
| Dynamic Processing Types | gate, compressor, expander, ducker, compandH, compandS |
| EQ | 4-band parametric |
| Channel Delay | 0–2,600 samples |
| Multi-effects Processors | 2 |
| Multi-effects Types | reverb, delay, modulation effects, guitar effects, dynamically controlled effects, combination effects |
| **I/O** | |
| Analog Inputs | (2) unbalanced RCA |
| Analog Outputs | (2) unbalanced RCA |
| Digital I/O | (1 pr.) S/PDIF (RCA) |

For manufacturer information, please see Contact Sheet, p. 227




# ACTIVATE
## YOUR SPACE

Introducing the new model 1029A. It's our latest active nearfield monitoring system that has all the integrity and performance you'd expect from a Genelec. It's accurate, features amazing dynamics, and can deliver big SPL's from a plug n' play, rugged aluminum enclosure. Best of all, the 1029A is *surprisingly affordable.* Then, there's our new 1091A, a matching, active subwoofer specifically designed to be the 1029's perfect bottom-end sonic companion.

And because Genelec has 20 years of active monitor design and manufacturing, you can be sure what you hear represents the audio truth. So, regardless of your Space, large or small – or your professional persuasion – broadcaster or rock n' roll, this Genelec active monitor system will let you hear the trees – even if you're not in the forest.


**GENELEC®**

ACTIVE MONITORING

The Whole Truth And Nothing But The Truth.

In the U.S. please contact: Genelec Inc., 7 Tech Circle, Natick, MA 01760 Phone: 508/652-0900 Fax: 508/652-0909
International enquiries: Genelec, Olvitie 5.FIN-74100 Iisalmi, Finland, Phone +358-17-813311 Fax +358-17-812267 Web: http://www.genelec.com

**circle #594 on reader service card**

all connected outputs—a useful feature for troubleshooting connections outside the computer.

Using *Cubase*'s .pdf-format help, I was able to quickly configure the software for the DSP Factory. It's apparent that Steinberg put some thought into walking users through this process. For example, to designate the DS2416 as the recording hardware, the documentation directs you to simply pull down the Audio menu, select System, and make sure the ASIO driver is engaged. Then you press the ASIO Multimedia Setup button, and a pop-up menu allows you to select your type of I/O configuration (for example, Yamaha SW-1000XG, one DS2416, two DS2416s, and so on).

*Cubase* even comes with two DSP Factory Songs, which contain no audio but can be used as templates for creating projects. A 16-channel template and a 32-channel template are designed to expedite the setup process of each new project when using the DSP Factory. The inclusion of these preconfigured projects is a godsend—it saved me the hassle of redefining basic project information, signal routings, and I/O settings each time I wanted to start fresh.

## THE CUBASE CONNECTION

When used with *Cubase*, the DSP Factory works in tandem with the VST architecture. The documentation suggests that you view the DSP Factory as an external digital mixer, with signals being fed to and from VST (although you can bypass the VST architecture completely and use only the DS2416 mixer). Tracks and inputs may be processed either by the DSP Factory (where they are referred to as DS channels), by the VST system (called VST channels), or by a combination of both. Most of the DS2416's mixer functions can be automated alongside the VST channels, as well.

*Cubase* offers dedicated DSP Factory windows that you access from the Audio menu. Here you can select the Input Console, Channel Overview, Bus/Aux Console, FX Editor, and Output Patchbay. It's important to remember that the VST mixing system remains active even when you're working with DS channels. So if you plan to process a channel with the DSP Factory, you'll need to maintain unity gain of the corresponding fader in the VST mixing window. Working with two mixing surfaces at once took a little getting used to, but I was able to adjust.

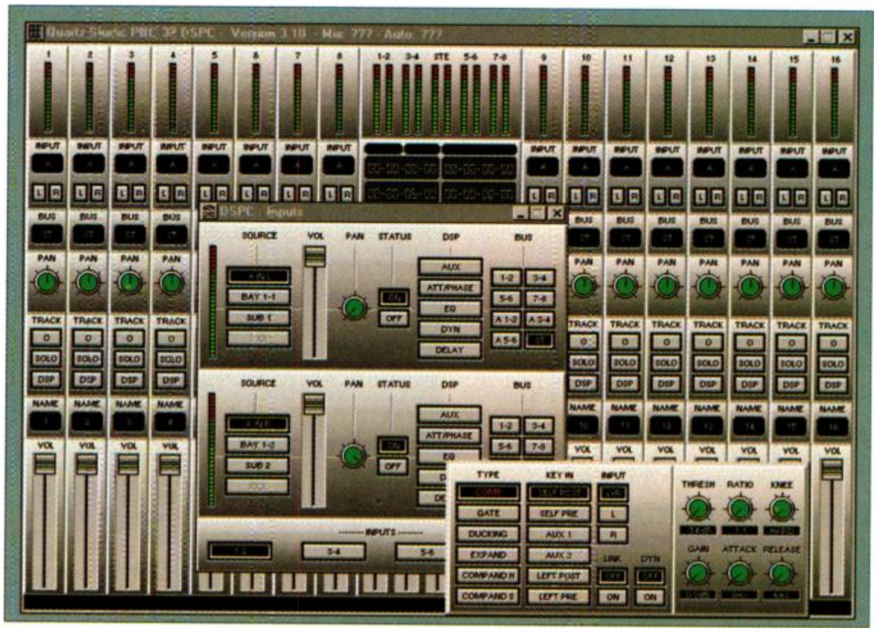
The Input Console is the main work area for the DSP Factory. It displays the 24 input faders; their corresponding meters, pan controls, 4-band parametric EQs, and dynamics processors (either gate, compressor, expander, ducker, compander hard, or compander soft); the stereo bus module (which also includes a 4-band parametric EQ and a dynamics processor); and global controls such as automation Read and Write. The Input Console is also where you activate bus and aux sends and select input sources for the DS channels.

Although you can change input routings from the default mode (which engages only the DS2416 I/O, not the AX44), there are restrictions as to what can be routed where. For example, DS channel 1 can receive signals only from input 1 of the AX44 or from VST channel 1. The process of assigning inputs from the two AX44s had me a little baffled; it took some time to find the appropriate information in the *Cubase* documentation.

engines are hard-wired to aux sends 5 and 6, and can be accessed through the FX Editor window. A pop-up menu allows you to select either FX Unit 1 or FX Unit 2, and another lets you choose one of the 40 preset effects programs. Once an effect is selected, its parameters are displayed in the window.

The Bus/Aux Console displays master levels, meters, and mutes for the six aux sends and eight buses. The Output Patchbay allows you to route these signals (and the stereo bus) to any of the DS2416 or AX44 outputs. The right-hand side of this window also displays sample-rate settings.

Overall, *Cubase* and the DSP Factory are a very impressive combination: the logically designed software makes readily apparent the power of the hardware (in particular, its routing capabilities). I did encounter some sputtering during playback, especially when scrolling the Input Console window; in all fairness, though, this was probably due to the limited RAM (32 MB) installed on the test computer. Again, you'll need to check the system requirements of your particular software; although 32 MB is generally adequate, 64 MB or more is often needed for flawless playback.



**Canam Computers' *Studio Pro 32* has dedicated DSP Factory control with full support for all DS2416 functions. Displayed here are the Mix window, Input Console, and Dynamics Processing window.**

## INDUSTRIAL DSP

After finding my way around the software, I decided it was time to put this puppy to the test. I started by listening to one of the demo songs provided by Yamaha. Everything sounded great—clearly, Yamaha had put a lot of care into recording these tracks, and none of them required much EQ or dynamics processing. So I imported some not-so-pristine tracks into the system through the DS2416's digital I/O. (I figured I'd save the demo tracks for checking out the multi-effects processors.)

The first track I imported was a stereo drum pair, which I have always considered the most challenging test of any parametric EQ; the frequency response of a drum mix covers almost the entire spectrum. After tweaking the drum tracks for only a few minutes, I determined that the EQ of the DSP Factory sounds exactly like that of an 03D: warm but not muddy; clear but not brittle.

The dynamics processors also do not disappoint. To evaluate the compressors, I used a very dynamic vocal track (I'm talking a 50 dB dynamic range here), again imported from my own hard drive. Although digital dynamics processors offer more detailed parameter control than their analog counterparts, dialing in effective attack and release times can be a bit tricky—there's something to be said for just twisting a dial on an analog box. However, there's also something to be said for the DSP Factory's compressors: they behave *a lot* like analog units. I was able to gain dynamic control over the vocal track quickly and effectively, and without generating that all-too-familiar overcompressed pumping sound.

I then imported a rather sloppy snare drum track (one with a lot of stick drag) to test out the gate. Homing in on the proper threshold was a breeze, and the result was a very tight-sounding snare. The expander also worked wonders on cleaning up an electric-guitar track that was plagued by a loud amplifier hum.

In all probability, you'll never use the DSP Factory's ducker or either of its companders, but having them available means your studio is equipped to tackle the most challenging projects. I tested them all and was particularly impressed with the hard compander's performance on an acoustic-guitar track: I was able to eliminate a rather annoying string buzz with it. All in all, these are very powerful dynamics processors, comparable to certain high-end outboard gear.

The best part about the DSP Factory is that all available processing—the delays, EQs, dynamics processors, and multi-effects processors—can be used simultaneously without compromising any of the effects. A lot of new hardware products offer this feature, but Yamaha has really delivered it here: at one point I had 16 channel EQs, 12 dynamics processors, 4 delays, and 2 multi-effects programs engaged all at once, and I didn't hear a single artifact from any of the processors. Now *that's* power.

## AX44 I/O Specifications

| | |
|---|---|
| **Analog Inputs** | (4) ¼" TRS |
| **Analog Outputs** | (4) ¼" TRS |
| **A/D Conversion** | 20-bit, 128x oversampling |
| **D/A Conversion** | 18-bit, 8x oversampling |
| **Digital I/O** | N/A |
| **Other Connections** | ¼" headphone jack |

### SIGNS OF EFFECTION

The DSP Factory comes loaded with multi-effects. As with the 03D, all its algorithms have been borrowed from the company's acclaimed REV500 processor. There are 40 effects types, all of which are available from both of the DSP Factory's multi-effects engines.

Included are an array of reverbs (Hall, Room, Stage, Plate, Gated, Reverse); delays (Mono, Stereo, Modulated, Tap, Echo); modulation-type effects (Chorus, Flange, Symphonic, Phaser, Auto Pan, Tremolo, Rotary, Pitch-Shift); guitar effects (Distortion, Amp Simulation); dynamically controlled effects (DynaFilter, DynaFlange, DynaPhaser); and combination effects.

I was particularly impressed with the Room Reverb. I shouldn't have been so surprised, considering I've always been a fan of Yamaha reverbs. But when I engaged the effect on a lead vocal track, I couldn't help but smile: the same track on another digital audio workstation required that I use two reverbs and a tap delay. With the DSP Factory, the Room Reverb was all that I needed.

I also liked the system's Symphonic Chorus. This effect sounds much fuller than a regular chorus, and it fattened up some background vocal tracks immensely. The dynamic effects are also very cool, as are some of the combination effects.

My only complaint about the DSP Factory's effects is that, to my ears, the guitar distortion wasn't particularly convincing. Then again, I have yet to come across a software-based amp simulator that I do like. Also, the system's multi-effects processors can accept only mono input signals, so true stereo reverb is out of the question. Perhaps Yamaha will address this limitation and provide an upgrade in the future.

### ADEPT A/D/A'S

To evaluate the quality of the AX44 converters, I performed some tests by miking an acoustic guitar. My monitoring chain included a pair of Genelec 1031As, with signal delivered through a pair of gold-plated Canare cables.

First I put up a Shure SM57 mic and routed it directly into one of the AX44's mic/line inputs. Playback revealed a decent tone, although it didn't convince me to trade in any of my high-end mic preamps. Still, the sound was usable and a lot better than what I've heard from some other products' onboard preamps.

I then used a more professional combination of a Neumann U87 with a Drawmer 1960 tube preamp/compressor—a setup I typically employ for tracking guitars, acoustic pianos, and vocals—and routed the signal into one of the AX44's line-level inputs. The guitar tone was rich and detailed, and the converters perfectly captured the sound of the Neumann/Drawmer combo. I also recorded a vocal using the same gear, and the result was another warm and clear track that required very little processing in the mix.

With 24-bit recording conversion now the industry standard, the ante has been upped for every manufacturer. With that in mind, I wasn't expecting much from the AX44's 20-bit converters. Was I ever surprised! I had a hard time differentiating them from the 24-bit converters that I use in my own studio. Granted, I'm not comparing the AX44 converters to those in the Apogee AD-8000, but they sound impressive for what they are. I would have no reservation about using them in any professional situation.

One issue I need to mention, though, deals with the CPU-mounting nature of the AX44. I'm not a big fan of having analog audio anywhere near my computer, let alone inside the chassis. A CPU is a volatile place, with the potential for generating electronic interference. I didn't encounter any sonic aberrations due to this design, but that's not to say such a problem will never pop up. (Yamaha is considering making external I/O modules available in the future.)

## MIDI MAYHEM

Just when I thought things couldn't get any cooler, I installed the Yamaha SW-1000XG PCI Audio MIDI Card. The SW-1000XG is essentially a Yamaha AWM2 tone generator coupled with a 12-track digital audio recorder, more DSP effects, and *XGWorks* (the SW-1000XG's proprietary digital audio recording and MIDI sequencing software.)

The SW-1000XG delivers 64-voice polyphony and offers more than 1,200 different voices derived from 20 MB of sample ROM. It's compatible with General MIDI and Yamaha XG sounds; Yamaha VL, VH, and DX sound generation is available through the company's optional PLG100-series expansion boards. The card provides stereo analog I/O on an ⅛-inch jack, S/PDIF I/O on RCA jacks, and a breakout cable with MIDI In and Out jacks.

*XGWorks* has MIDI sequencing and multitrack audio recording features, in addition to waveform audio editing, digital signal processing, and basic mixing capabilities. While *XGWorks* can be configured to take advantage of the DSP Factory's I/O, the software doesn't support the DS2416 mixing functions. What you can do is route digital audio from the SW-1000XG to the DS2416 (and vice versa) by connecting the two cards with a serial cable. Your controlling software will determine exactly how the signals are routed into and combined with the DS2416's audio tracks. (With *Cubase*, eight channels of audio from the SW-1000XG can be routed into the DSP Factory via DS channels 9 through 16 or 17 through 24.)

Although a complete review of the SW-1000XG and *XGWorks* is beyond the scope of this article, suffice it to say that using the card with the DSP Factory opens a lot of creative doors, bringing additional audio capabilities and MIDI into the system. The two devices make an impressive pair.

## HARD COPY

The provided DSP Factory documentation is sparse, but it does a good job of explaining how to install the various system components and provides basic setup diagrams for novice users. For more elaborate explanations of the DSP Factory's functions, you'll be instructed to refer to the documentation provided with the controlling software. (Again, I found Steinberg's literature to be comprehensive and informative.)

At its reasonable price, it's hard to beat the DSP Factory—heck, it's hard to beat at any price. The processing is top-notch, the mixer and recorder are extremely powerful, the A/D/A converters are solid, and the whole thing sounds great. Coupled with some of today's best software, the DSP Factory can transform virtually any PC into a powerful digital audio workstation. Look out, everyone—Yamaha is about to corner the market again. ❁

# CAKEWALK

## PRO AUDIO 8 (WIN)

*A host of new high-end features add more flavor to an already delicious cake.*

By Scott R. Garrigus

Until my review copy of Cakewalk *Pro Audio* 8 arrived, I had been a happy user of version 6 of that program. I was aware that version 7 had been released, but the company rolled out version 8 before I even got around to buying it. *Pro Audio* 8 offers many new features, including 32-bit floating-point stereo audio effects, MIDI effects, nondestructive pan and volume audio envelopes, and the ability to view video files. Support for 24-bit, 96 kHz hardware also makes this release appealing.

As soon as I liberated my new software from its cardboard package, I installed it on my 300 MHz Pentium II PC with 64 MB of RAM, a Sound Blaster–compatible sound card, and an Event Electronics Gina 20-bit audio interface. Installation went without a hitch, and I was up and running within ten minutes.

### HOW DOES IT SOUND?

The first thing I wanted to determine was how much the new support for high-end audio hardware improved the sound on my system. The Gina sound card provided up to 20-bit, 48 kHz audio, so I couldn't check out *Pro Audio* 8's full range. Nevertheless, I had an awesome time recording audio at a rate higher than CD quality. And yes, I did notice a difference in the quality of the sound. The higher specs yielded a more detailed and spacious sound, and I was better able to pick out distinct parts in a dense mix.

Even my Sound Blaster card sounded better than ever before. How is that possible? It supports a maximum of only 16-bit, 44.1 kHz audio, but thanks to *Pro Audio* 8's new audio mix engine, audio is processed internally with 32 bits. This provides more efficient processing, and with the program's built-in output limiter, the audio is soft clipped before being output at the appropriate bit depth and sampling rate. So even though the Sound Blaster offers only CD-quality specs, the final output sounded crisper and more detailed than what I'm accustomed to hearing.

*Pro Audio* 8 supports 16-, 18-, 20-, 22-, and 24-bit recording along with common sampling rates of up to 96 kHz. However, as with other digital audio sequencers, you can have only one resolution and one sampling rate per project, which means that all tracks in a project require the same settings.

*Pro Audio* 8 gives you several options for handling files of different formats. You can use the Tools/Change Audio Format command to switch a project's default resolution between 16- and 24-bit. The program automatically converts imported WAV files to the current project's sampling rate. If you need to convert a project to another sampling rate, simply copy all of the data to the clipboard, and paste it into a new project with a different sampling rate.

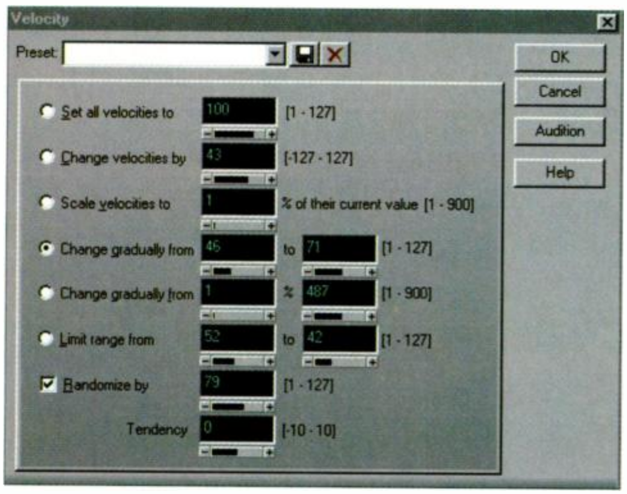



**FIG. 1:** ***Pro Audio* 8's new MIDI effects plug-ins allow you to make changes to your MIDI tracks nondestructively and in real time. The Velocity effect displayed here contains parameters that are not available from the standard Edit menu.**

### NEW AND BETTER EFFECTS

Another benefit of *Pro Audio* 8's audio engine is improved real-time plug-in effects. Audio effects are now processed with 32-bit floating-point accuracy, enabling them to run faster (particularly on a Pentium processor) and sound better. Most effects can be processed in stereo. *Pro Audio* 8 ships with 12 DirectX audio plug-ins, including Parametric EQ, Pitch Shifter, Delay, Chorus, Flanger, Reverb, and Time/Pitch Stretch. They all sound good, especially the EQ and Reverb. The Time/Pitch Stretch effect includes a formant-preserving option, but it still needs some work. Any pitch change in excess of +/-1 semitone, with the formant preservation option selected, produces noticeable artifacts.

The real-time MIDI effects are totally new and very cool. These proprietary (not DirectX-compliant) plug-ins work like the audio effects but on MIDI tracks instead. They can be used in real time or as offline processes. The MIDI effects include Arpeggiator, Chord Analyzer, Echo Delay, MIDI Event Filter, Quantize, Transpose, and Velocity.

Keep in mind that these are MIDI processes, not audio processes. As with all MIDI delays, Echo Delay will use additional voices of polyphony if echoes occur while the original note is still playing.

At first, I wondered why I would need these plug-ins if similar functions were already available from the Edit menu. I realized that the plug-ins are a thing of beauty when used in real time. Whereas the Edit menu functions work only in a destructive manner, the real-time plug-ins are nondestructive. This allows you, for instance, to easily experiment with different quantize settings as you listen to the music without altering the original data. Very useful, indeed. Moreover, the real-time MIDI effects add features not included in the standard effects. For example, the Velocity effect has a number of adjustable parameters that you won't find duplicated elsewhere (see **Fig. 1**).

I found all the MIDI effects to be quite useful except, perhaps, the Chord Analyzer, which recognizes chords only when they're actually played as chords. You can't, for example, analyze all the notes in a measure of music to find the common chord; that would be a very handy feature.

### EASIER THAN EVER

After my fascination with the superior sound specs subsided a little, I investigated the improved user-interface:

For manufacturer information, please see Contact Sheet, p. 227

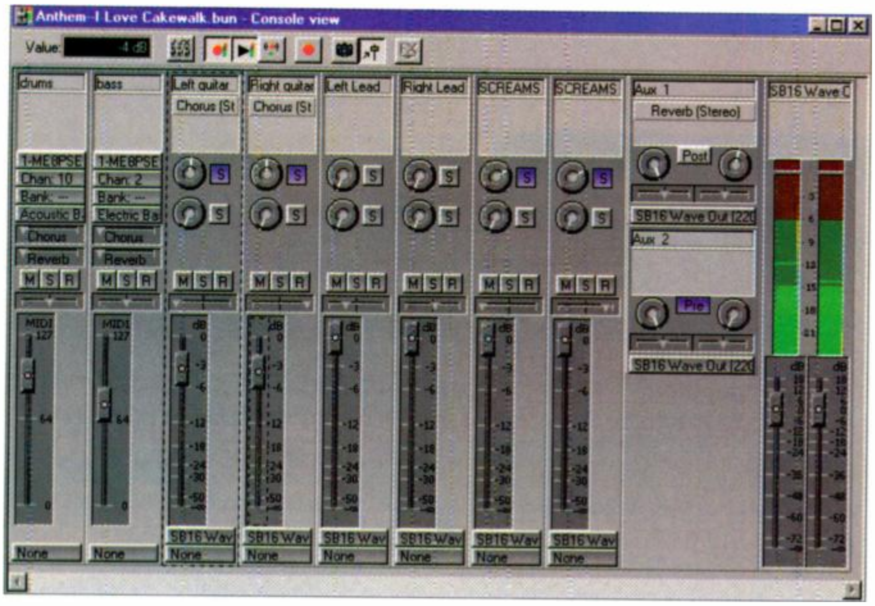



**FIG. 2: The enhanced Console view is Pro Audio 8's answer to the virtual mixer. From this view you can record and mix all your music just as you would with a real mixing console. This view also makes assigning real-time effects much more intuitive.**

it's both familiar and a bit foreign. The most noticeable changes are the new toolbars. *Pro Audio* 8 now includes dockable toolbars containing shortcuts for all the program's basic functions, including file operations, loop settings, marker settings, song position, and transport. One thing that I'm sure we'll all miss is the familiar screaming face icon that once adorned the onscreen Panic button. It's been replaced with an exclamation point.

The new toolbars contribute to *Pro Audio* 8's ease of use. For example, it's now a snap to create loop regions: simply start playback and click on the Insert Marker button to drop a marker at the start of the region and another at the end. Next click on the Markers View button, and select the first marker on the list. This highlights the region between the two markers. Finally, click on the Set Loop To Selection button and the Loop On/Off button. That's all there is to it.

The zoom functions also are simplified. In addition to the usual zoom-in and zoom-out buttons located in the lower-right corner of each view window, the program now has Slider buttons. Click and hold down the Slider button to reveal a pop-up slider that you can easily adjust by moving the mouse. *Pro Audio* 8 features a new Lasso Zoom button, as well. Click on this button, and then drag the mouse over the data to create a selection. As soon as you let go of the mouse button, the new zoomed data fills the screen. One feature I'd like to see is a Fit-to-Window zoom function that would provide a quick overview of your entire file. However, *Pro Audio* 8 does offer some other useful tricks, such as the Zoom Undo function, which returns you to your previous zoom level. (Each zoomable view has its own zoom history.)

The added ability to open multiple project files simultaneously is helpful. Now you can easily swap parts from one project to another by cutting and pasting or dragging and dropping. Because the File/Merge and File/Extract commands are no longer needed, they've been removed. Each open project also has an unlimited number of undo levels and its own unique undo history. This feature brings *Pro Audio* 8 more on par with the way a true Windows interface should be. In fact, the program is more Windows-like than ever. Track selection now works just the way file selection in Windows does. Click on a track to select or deselect it; drag over several tracks to select them all. Context-sensitive right-click pop-up menus are now available in all views, as well. This makes working with the program much more intuitive.

## CONSOLING VIEW

The most significant change to the user interface is the Console view (see **Fig. 2**), which was first introduced in version 7. *Pro Audio* 8 offers additional enhancements to this "virtual mixer." If you already have StudioWare (Cakewalk's customizable control interface), why would you need the Console view? Well, although you can create a simple virtual mixer with StudioWare, the program is best suited to controlling outboard MIDI gear. The Console view, on the other hand, provides real-time effects inserts, aux buses, and audio metering, in addition to the usual faders and pan knobs. In fact, you can use the Console view to control all of your tracks during the recording and mixing process, just as you would on a real mixing console.

Essentially, the Console view mimics the Track view: for every track present in the Track view, there's a corresponding mixer module displayed in the Console view. Both MIDI and audio tracks are represented, each with appropriate controls. The MIDI modules contain effects inserts for the MIDI real-time effects plug-ins; port, channel, bank, and patch assignment buttons; chorus and reverb controls; and the controls common to all modules—Mute, Solo, and Record buttons. There's also a Pan slider, a volume fader, and a recording-source assignment button.

The audio modules are similar, with effects inserts for the real-time audio effects plug-ins. But instead of assignment buttons or chorus and reverb controls, there are two aux sends for the two available aux buses. (You can easily configure *Pro Audio* 8 to use up to 16 aux buses, and the number of

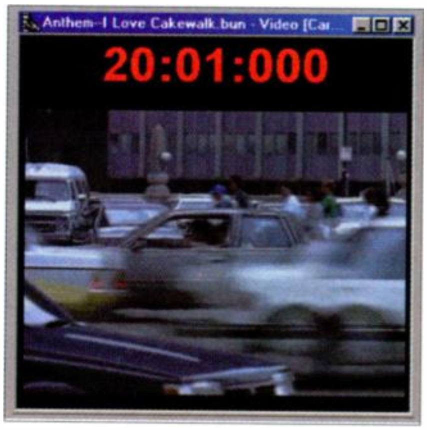
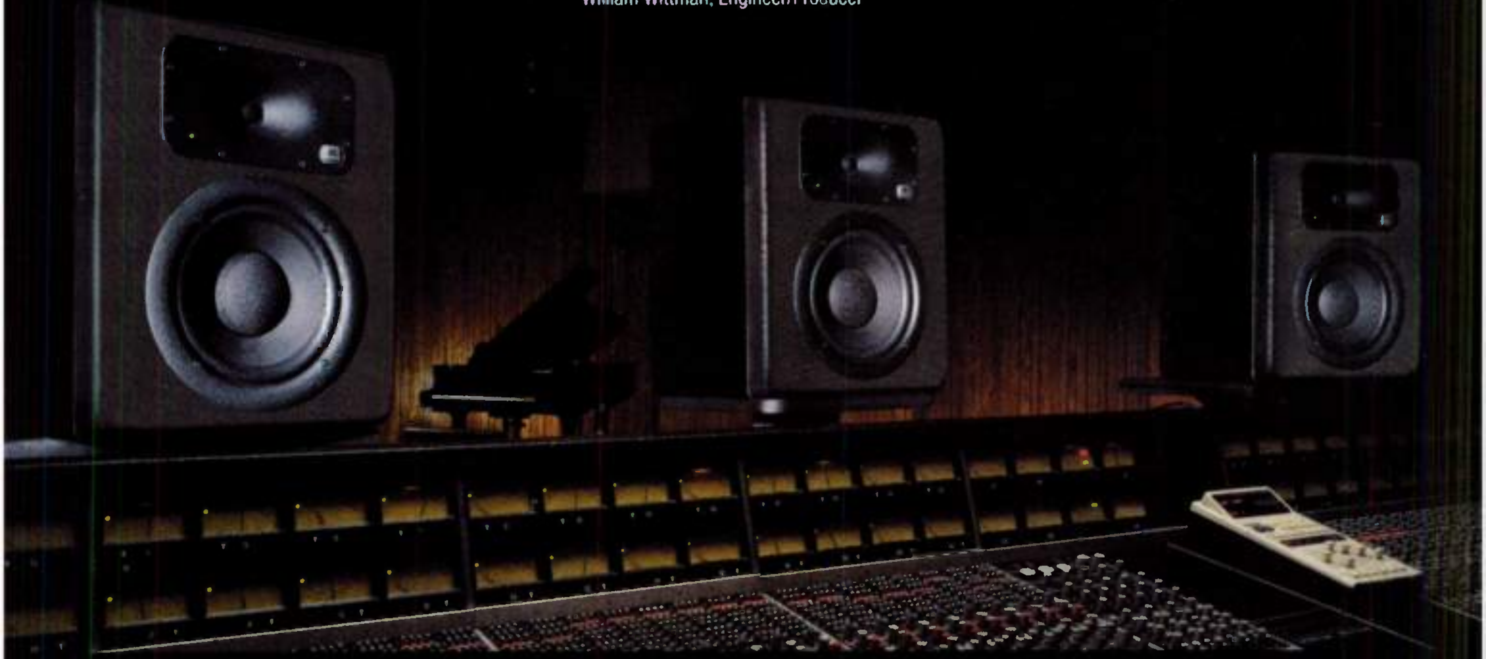



**FIG. 3: *Pro Audio* 8 now allows you to add music to video files. After you insert a video, it is played back along with your MIDI and audio tracks in perfect sync.**

# "When A $2,000 System Sounds As Good As A $50,000 System, I'd Say They Got It Right."

William Wittman, Engineer/Producer



## The First Integrated Professional 5.1 Monitoring System With THX® Approval

The All-New JBL LSR Monitors are, quite literally, just that. Highlighted by a long list of performance-tailored components and customer-inspired features, they're like no other systems on the market today. The entire line, including the LSR32 3-way, 28P 2-way and 12P Subwoofer, is a technical triumph; resulting in new standards and performance levels for a rapidly emerging multi-channel recording industry.

## Performance-Tailored Components

Revolutionary transducer designs, optimized network topologies and innovative materials are some of the reasons why the LSR line is being hailed as 'the world's most advanced monitor'. JBL's all-new ***Differential Drive***® woofer permanently dispels the notion that better linearity, higher power handling and greater dynamic accuracy are somehow an unobtainable, evil triangle. ***Dynamic braking*** produces truly accurate bass at higher SPL's with maximum reliability. Composite materials, including ***Carbon Fiber*** in the woofer as well as ***Titanium*** and ***Kevlar***® in the high and mid frequency components, insures performance that is always optimally maintained.

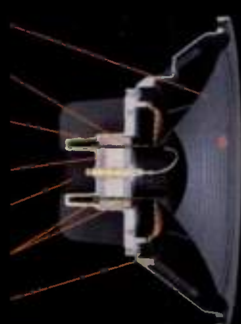
## Not Just A Better Spec... A Better Monitoring System

While all companies boast about their specifications, JBL went one step further. To guarantee that every component of the LSR family worked together for optimal performance, LSR development employed JBL's unique 'system-engineered' design philosophy. Simply put: the entire line was researched and refined as one, with an overall performance goal in sight. What this means to you is a monitor and subwoofer that work together as a system; delivering stunningly uniform and accurate performance in both stereo and multi-channel applications.

**LSR 32**
12" 3-way mid-field monitor with rotatable Mid/High Elements.

**LSR 28P**
8" 2-way close field monitor with bi-amplification and active filtering.

**LSR 12P**
12" Active Subwoofer with Bass Management System.




JBL PROFESSIONAL

H A Harman International Company

To find out more about the revolutionary LSR Monitors, visit JBL Professional at www.jblpro.com

SEE US AT NAMM BOOTH #439

circle #604 on reader service card

sends per channel strip will correspond to the number of aux buses.) Audio modules also have output assignment buttons.

Assigning real-time effects to tracks is much easier now with the Console view. The archaic and confusing Effects view is gone, thank goodness. Now you simply right-click in any track insert area, and the appropriate effects plug-ins are displayed for either MIDI or audio tracks. If you want to apply a set of effects to several audio tracks, you can set up one of the aux buses, which can be either pre- or postfader.

To help with your mixing chores, there are a number of basic and advanced fader grouping options. You can group faders to move in sync or to move in different ratios—one fader moves quickly as the other moves slowly, for example. You can also group them to move in opposite directions for smooth crossfades. Up to 12 groups can be created for faders and knobs, and you can have another 12 groups for buttons. This is great for controlling simultaneous aux send changes or muting a number of different tracks at once.

As with StudioWare, you can record your mix changes for later automation and have the onscreen controls move automatically to reflect the recorded automation during playback. A major problem with the Console view, however, is latency. During my tests, whenever I manipulated an audio control, such as a volume fader, it took two or three seconds before I heard the change at the output.

## OPTIMIZE YOUR AUDIO OPTIONS

Depending on the speed of your computer system, you may experience some latency when dealing with audio tracks in *Pro Audio* 8. You can reduce this unwanted effect, however, by adjusting the program's size and the number of queue buffer settings. These settings are found in the Tools/Audio Options/Advanced dialog box, but unfortunately, there is no universal set of optimum values. Every computer system is different, so you'll have to experiment. Here are some guidelines to help you along the way.

**Step 1.** Keep the Enable Read Caching and Enable Write Caching parameters deactivated. Unless you have an older computer system, these settings can hurt *Pro Audio* 8's performance.

**Step 2.** You can leave the File I/O Buffers and I/O Buffer Size settings to their defaults of 32 and 64 respectively. These settings are directly related to the speed of your hard-disk drive and have more to do with consistent playback than latency. If you get hiccups during playback, you may want to increase the I/O Buffer Size settings, but most probably the default settings will be fine.

**Step 3.** Set the Wave Queue Buffers to 3. *Pro Audio* 8 will probably be unable to commence playback if they're set to anything less. This parameter depends on the speed of your CPU, which is directly related to the latency issue. I found 3 to be a good setting for my system: anything more is essentially wasting memory, but if you have a slower system, you may need to increase the setting to 4.

**Step 4.** The Wave Buffer Size is also directly related to latency. The higher the value, the greater the latency; the lower the value, the lesser the latency. You must be careful, however, not to set it too low. A smaller buffer-size value also reduces the number of audio tracks you can play at once, as well as the number of real-time effects that you can use simultaneously. I've found 32 to be a good setting for my system; it allows me to have around 20 audio tracks playing at once, and the latency is very small.

**Step 5.** Leave the DMA settings for your sound card at their default. If you accidentally alter them, use *Pro Audio* 8's Wave Profiler features to have them automatically determined again.

That should do it. You should now have *Pro Audio* 8 pumpin' out audio tracks at a nice pace and be able to use the new Console view with very little delay between knob tweaks and actual change in output.

(MIDI tracks worked fine.) This was very frustrating.

When I contacted Cakewalk, I learned that *Pro Audio* 8 is shipped with default playback buffer settings to ensure reliable playback on most computer systems. These settings can be optimized to provide lower latency if you have a fast computer with a fast hard disk (see the sidebar "Optimize Your Audio Options"). After a bit of tweaking, I was able to reduce the latency on my system to a matter of milliseconds as opposed to seconds. I continued to notice a delay when recording fader automation, but it was less annoying than what the default settings provided. More importantly, no latency occured during playback.

## EVEN MORE INGREDIENTS

Two other exciting new features in *Pro Audio* 8 are video file support and pan and volume audio envelopes. Multimedia composers can now use *Pro Audio* 8 to add music to any AVI, QuickTime, or MPEG video file. Simply activate the new Video view, insert a video file using the right-click pop-up menu, and you're ready to record (see **Fig. 3**). The video is displayed in sync with your MIDI and audio tracks during playback. The Video view shows the video itself and a time readout that can be set to display measures, beats, and ticks; SMPTE time code; or frames. While you're navigating a project, the Video view stays in sync with the current cursor position. When you've finished recording your music, you can export both the combined video and audio data to a file. It's too bad that even though you can load three video formats, you can only export to AVI.

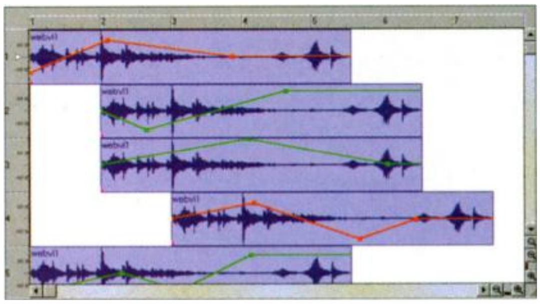


**FIG. 4: From within the Audio view, you can add pan and volume envelopes to any audio segment or group of segments. These envelopes allow you to make nondestructive changes to your audio as it plays back in real time.**

Not only does the Console view allow you to automate the panning and volume of your audio tracks nondestructively, but the new pan and volume envelopes let you manipulate those settings visually, as well. You can add envelopes to any audio event from within the Audio view (see **Fig. 4**). Simply right-click on a segment of audio and select the Envelope function from the pop-up menu.

Once assigned, you can manipulate an envelope by simply dragging the control points or lines. You can also add more control points with a quick double-click. By using Linked Clips, you can set envelopes for a number of different segments at once. Just select one of the linked clips in the Audio view, add and manipulate an envelope in one clip, and those settings are reflected in all the others. The only thing that disappointed me about this new feature is that it has no support for effects envelopes. That

**Pro Audio 8**
**Minimum System Requirements**
Pentium 100; 16 MB RAM and Windows 95/98, or 32 MB RAM and Windows NT

would be a nice touch. Perhaps it will be added in version 9.

## THE LAST BIT

Although I've mentioned all the major new features of *Pro Audio* 8, the program also has many smaller but significant additions. These include metronome enhancements, anchors for audio events, the Patch Browser, and the Record drop-down list. I would also be remiss not to point out the major enhancements to Cakewalk's *Musician's Toolbox III*, which ships with the Deluxe edition of the program. The *Toolbox* is a set of two CD-ROMs filled with brand new MIDI and audio files, utilities, digital video clips, and interactive tutorials that can help you get going with *Pro Audio* 8. The collection includes more than a gigabyte of multimedia files and a search engine to help you locate the type of files you need.

The program isn't perfect. Having the latency problem totally eliminated would be great, although I've experienced this problem when using other software, and eliminating it might be impossible, even with today's fast computers. I'd also like to see some of the additions incorporated that I mentioned earlier, such as effects envelopes, fit-to-window zoom, more video export formats, and a better Time/Pitch Stretch effect. Despite these few gripes, however, I believe *Pro Audio* 8 is an excellent first-time buy and a definite upgrade for those already using an earlier version. Cakewalk has clearly brought new meaning to the word *pro* in *Pro Audio* 8.

---

**Scott R. Garrigus** *has been a Cakewalker since the* Professional *version spawned* Pro Audio 8. *Before that he was doing all his MIDI sequencing on an Atari. Can you believe that? You can reach him via e-mail at scott@garrigus.com or via the Web at www.garrigus.com.*

**CAKEWALK MUSIC SOFTWARE**
Pro Audio 8 digital audio sequencer
$299
$399 for Deluxe Edition
with Musician's Toolbox III

| | |
|---|---|
| FEATURES | ■■■■ |
| EASE OF USE | ■■■■■ |
| DOCUMENTATION | ■■■■ |
| VALUE | ■■■■ |
| | 1 2 3 4 5 |

**PROS:** Improved system performance and interface enhancements. Powerful mixing options. Intuitive effects assignments via the Console view. High-end audio formats support. 32-bit internal processing. Real-time audio effects. Nondestructive audio pan and volume envelopes. Video file support.

**CONS:** Time/Pitch Stretch effect needs improvement. Video saved only in AVI format. Audio response latency problems. No effects envelopes.

**CIRCLE #438 ON READER SERVICE CARD**

# ANTARES

## ATR-1

### *Real-time pitch correction that really works.*

**By Rob Shrock**

Most electronic musicians keep their eyes and ears open for that magic "black box" that can perform miracles on lackluster audio. We've seen several such devices over the past few years: aural exciters, harmonizers, subharmonic synthesizers, amp simulators, even devices that let you remove the lead vocal from a favorite classic rock song and make a fool of yourself at a family reunion.

Of course, the "magic" of these devices is really innovative design and implementation, with intelligent compromises. Such is the case with the AnTares ATR-1 intonation processor. Known for its excellent *Auto-Tune* software plug-in (see the sidebar "Auto-Tune"), AnTares has developed a hardware version of the same pitch-correction technology and packaged it in an easy-to-use, single-rackspace black box that operates in real time. And it is indeed magical.

The front panel includes a two-line, 40-character alphanumeric LCD, a large data-entry knob, and two cursor-navigation buttons. Also on the front panel are buttons for selecting Program, Song, and System Edit mode and another button to cycle through pages within the selected Edit mode. A Bypass button does exactly what you'd expect. A vertical row of six LEDs indicates the input-signal level, and a horizontal row of ten LEDs indicates the amount of pitch correction at any given moment.

The back panel sports a 7-pin DIN AC power-cord receptacle, a MIDI In port, and a ¼-inch footswitch input. The mono, balanced, line-level audio inputs include ¼-inch TRS and XLR connectors, but they can't be used simultaneously to mix two signals. The line-level output includes ¼-inch TS (unbalanced) and XLR (balanced) connectors. A grounding switch lets you select circuit-board or chassis ground, which is a nice touch.

## DEFINING THE PITCH

The 52-page manual covers the basics of pitch detection and provides thorough information on the operation of the ATR-1. I strongly recommend that you read the manual; it's a short read, and it covers some concepts you must understand to operate the ATR-1.

The ATR-1 introduces a short processing delay (1 to 4 ms) that allows it to calculate the pitch of an incoming periodic waveform. Often, the algorithm begins calculating the pitch of the waveform before the amplitude reaches an audible level, but the delay ensures a stable and accurate pitch-detection process, which operates continuously. This delay is unnoticeable unless you run the ATR-1's output alongside the unprocessed input signal, in which case you hear a small amount of phasing. In practice, the delay is no problem.

However, the ATR-1 can process only certain types of signals. The input waveform must be periodic and free of excessive noise. Obviously, this limits the pitch correction to monophonic instruments and solo voices. Unison ensembles (such as a unison choir or violin section) cannot be processed because they have no clearly defined fundamental due to the minor pitch discrepancies within the ensemble. As long as you are dealing with signals like a lead vocal or solo sax, the ATR-1 can work wonders to correct intonation problems.

Some monophonic signals, such as breathy vocals, can cause pitch-tracking problems. Fortunately, the Sensitivity control helps in this regard; at the correct setting, this control allows even signals with quite a bit of bleed from other instruments to be pitch-corrected. For anything other than a clean, isolated track, the Sensitivity control is one of the crucial parameters to have at your fingertips.

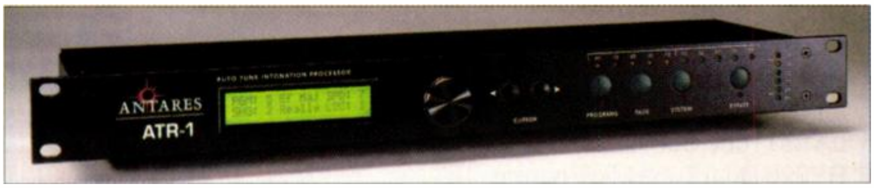
## PROGRAM MODE

The ATR-1 continuously tracks the incoming pitch of a live or recorded signal and compares it to the notes in a user-defined scale. If the signal matches a pitch in the scale, it is left alone. If there is a discrepancy, the pitch is shifted to the nearest scale tone. You can also specify certain notes in the scale to be bypassed, in which case nearby pitches in the input signal are not altered.

Scales are stored in Programs, of which there are 50 in the ATR-1's memory. The first 13 Programs are preset with the 12 major (and relative natural minor) scales and the chromatic scale, but you can save different scales in these locations if you want to. Unlike the plug-in versions of *Auto-Tune*, the ATR-1 can deal only with 12-tone equal-tempered scales; it has no microtonal capabilities.

Each Program shifts all pitches that are not in the selected scale, so "accidental" notes will be altered as they pass through the device (except notes that are set to be bypassed). I typically used the chromatic setting because most of the tracks I processed weren't strictly diatonic, and they had relatively good pitch to begin with. If you're working with a diatonic melody that stays in one or two keys, the corresponding Program might work well for you.

Another vital Program parameter is the Speed setting, which determines how quickly the pitch is adjusted toward the scale tone. Fast Speed settings work well on instruments that change pitch quickly without a lot of inflection,




**The ATR-1 intonation processor is easy to use and quite effective at correcting pitch problems in monophonic signals. However, it has no digital I/O or analog level controls.**

such as oboes. Slower settings allow more expressive pitch gestures, such as vibrato and slides. You can also program the ATR-1 to change the Speed in response to a user-selectable MIDI Control Change message.

Setting this parameter correctly is crucial. If it's too fast, any natural vibrato will be completely removed, making the part sound mechanical; if it's too slow, the pitch will not be corrected to the target note before the next note begins. A value in the range of 6 to 12 generally corrects the pitch without tampering with the vibrato and other gestures, leaving a natural-sounding and in-tune performance. When you find the right setting, the result is gorgeous. However, the normal tendency is to use too fast a setting in an attempt to maximize the pitch correction (more on this later).

The ATR-1 also lets you introduce artificial vibrato into the signal with control over the waveform, rate, depth, and onset delay; the rate and delay can even be controlled in real time with user-selectable MIDI Control Change messages. If you must squeeze the life out of a part for the sake of pitch, you can at least add some modulation back to the signal. But unless you're going for an unusual effect, this isn't something you would do routinely, because artificial vibrato never sounds as good as the real thing.

If you need substantial pitch correction but don't want to remove the character of the performance, a viable alternative is to process the real trouble spots of the track in a second pass, using fairly slow settings for both passes. In many cases, this will do the trick without ruining the overall performance and removing all the desired inflections. Of course, this works only for recording applications; you can't do two separate passes in a live performance.

Each Program has its own Scale, Speed, and Vibrato settings. You can step through the Programs with a footswitch or by using the MIDI Data Increment message. This mode is intended primarily for applications in which only one or two scales are required, as is the case with highly diatonic music.

## SONG MODE

In addition to Program mode, the ATR-1 provides a Song mode. Up to 20 Songs can be stored in memory, and each Song can include up to 15 Steps. Each Step contains a Program or one of several other navigation items (more in a moment), along with its own Speed and Vibrato settings, which override the associated Program's settings. This lets you use the same scale in different Songs without having to save multiple copies of the Program with different settings.

Song mode is designed for live situations, where you might need to step through several Programs in a performance. You can cycle through the Steps in a Song with a footswitch or MIDI Data Increment messages. (You can also step through the Songs in memory using the same techniques.)

As mentioned earlier, each Step can include one of several navigation items instead of a Program. For example, when you activate a Step set to Bypass, the signal passes through the ATR-1 without correction. If the active Step contains the Loop item, the Song immediately jumps back to Step 1; this is

### ATR-1 Specifications

| | |
|---|---|
| **Frequency Response** | 10 Hz–20 kHz (+0.06/-0.23 dB) |
| **Sample Rate** | 46.875 kHz |
| **Data Format** | 20-bit linear, 56-bit internal processing |
| **A/D Converter** | 20-bit |
| **D/A Converter** | 24-bit |
| **Dynamic Range** | 103 dB (A/D), 105 dB (D/A), A weighted |
| **Inputs** | (1) XLR balanced; (1) ¼" TRS balanced |
| **Outputs** | (1) XLR balanced; (1) ¼" TRS balanced/unbalanced |
| **Display** | 2-line x 20-character LCD |
| **Dimensions** | 19" (W) x 1.75" (H) x 5" (D) |
| **Weight** | ATR-1 4 lbs.; power supply 1.3 lbs. |

For manufacturer information, please see Contact Sheet, p. 227

useful when you want to repeat a harmonic structure several times. (To exit the loop, you must select the next Song with the footswitch or a MIDI Data Increment message.) The Link item causes the ATR-1 to jump to the next Song in memory, which lets you create a sequence of more than 15 Steps.

## A BIT TOO PERFECT

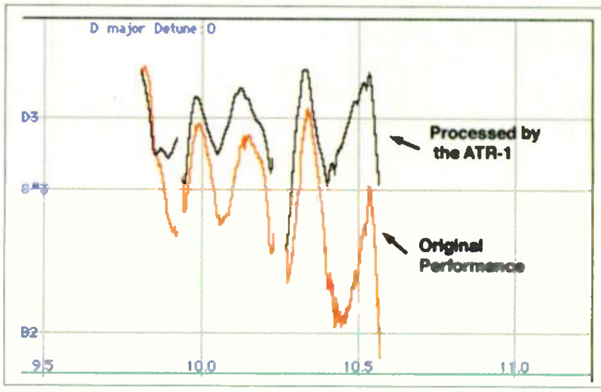
My first ATR-1 experience was with a prototype loaned to me by Ted Perlman, who owned one of the five that existed at the time. (With the exception of a few parameter tweaks, the production model I later received for review is identical to the prototype.)

I was producing a song for Dionne Warwick's current album, and I was faced with a tight deadline to finish the master and hand it over to the mix engineer. Warwick and I had tracked her lead vocal and 12 background parts (triple-stacked four-part harmony) in one very late night at the end of a long string of recording and concert dates.

Seasoned professional that she is, Warwick got all her vocals down in one evening, but we did not have the opportunity to touch up any parts later with fresh ears. After sorting through the various background parts the next morning, I found some places in the tripled background vocals where the deviations between like parts were too wide for my taste. They weren't necessarily out of tune, but rather out of focus.

I didn't have time to get involved in a *Pro Tools* session for vocal editing and comping because I also had some keyboard overdubs to do and a plane to catch that evening. The vocals had to be right, but they also had to be dealt with quickly. Enter the ATR-1.

Working with David Crigger at his project studio in North Hollywood, California, I ran each vocal track through the ATR-1 and made pitch corrections while transferring from ADAT tapes into *StudioVision* for comping and editing. Unfortunately, there is no digital input or output on the ATR-1, so we had to run each track through several conversions: from the ADAT's D/A converters into the ATR-1's A/D and out of the ATR-1's D/A into another set of A/D converters on the way to a Power Macintosh 7100. Another




COURTESY OF ANTARES

**This graph depicts the pitch of the original signal and the corrected signal. The ATR-1 is set to a D major scale with C# and B set to "Blank," thus allowing the expressive inflections to pass through without being corrected.**

drawback is that it has no I/O level controls. But the show had to go on.

A quick read through the manual got us up and running in about 15 minutes. Once we set the main parameters—Speed, Sensitivity, and Scale—we were in business. As mentioned earlier, my natural tendency was to overprocess the signal using a Speed setting that was too fast on the first couple of vocal tracks. These tracks had to be redone later in the session when it became apparent that the cumulative results were unnatural sounding.

We also noticed a low-level grounding hum polluting the ATR-1 output. The ground switch on the back panel did nothing to alleviate this problem, so we had to resort to using a two-prong ground-lift adapter. This eliminated the hum, and the ATR-1 output was virtually dead quiet. (The hum turned out to be a problem in the studio's electrical system, as subsequent experience demonstrated.)

Once we settled on a Speed setting that worked (8 to 10, depending on the track), it was easy to run a whole track through the ATR-1 and move on to the next take. If a section sounded fine, we just hit the Bypass button until we needed to use the pitch correction again. Ideally, I would have preferred to go bit by bit and process only certain sections of each track, varying the Speed parameter and diligently comparing it to other tracks. However, this would have taken many more hours, and pressing schedules demanded that we finish promptly.

I was dismayed that the ATR-1 has no digital I/O, but in spite of that, I was pleasantly surprised at the results. There was very little coloration in the sound from the ATR-1, although I could tell that each track had seen a few too many converters along the way for my liking. A minute amount of high-end noise and distortion was audible, but only when we mercilessly compared the processed sound to the original. Of course, no one listens to music this way, so the results were more than acceptable—they were brilliant! I was pleased, and the record company loved the track. End of story.

Well, almost. It's funny how pitch perception is relative. On the mix date, Warwick noticed that the background vocals had been polished, and she thought the tracks sounded a bit too "*Pro Tool*'ed." She asked the mix engineer to blend a little of the raw background tracks in with the pitch-edited parts, and then she was happy, too.

This brings up an interesting effect I've played with in the past: mixing a pitch-corrected signal with the raw signal can sometimes make a great doubling effect, especially with a little delay on one of the channels. It takes some experimentation to determine which channel (raw or pitch-corrected) should

**ANTARES**
ATR-1 intonation processor
$1,199

FEATURES
EASE OF USE
AUDIO QUALITY
VALUE
1 2 3 4 5

**PROS:** High-quality pitch correction of monophonic sources. Simple to use. Preserves performance character and inflection while correcting pitch. Very little sonic coloration. Custom scales. Can be controlled via MIDI or footswitch.
**CONS:** No digital I/O. No analog I/O level controls. A bit expensive.
CIRCLE #439 ON READER SERVICE CARD

## AUTO-TUNE

AnTares Systems originally implemented its pitch-correction algorithm as a TDM plug-in called *Auto-Tune* ($599). This plug-in is currently available in a number of incarnations, including VST ($399), DirectX ($299), MAS ($399), and V8 ($599), as well as a stand-alone version for Macintosh that uses the computer's digital hardware ($399).

*Auto-Tune* operates in one of two basic modes: Automatic or Graphical. In Automatic mode, *Auto-Tune* functions in virtually the same way as the ATR-1. After a scale is specified, the Retune slider sets the speed of pitch correction, and the Tracking slider is similar to the ATR-1's Sensitivity setting. The Vibrato parameters are also the same as those in the ATR-1.

In Graphical mode, the Retune slider has the same effect as in Automatic mode—determining how quickly pitch correction is applied—but the target pitch is not a scale tone; it is determined graphically by the user. As the signal is played into *Auto-Tune*, a waveform display at the bottom shows the current audio section, whose pitch profile is displayed in the main window (see **Fig. A**).

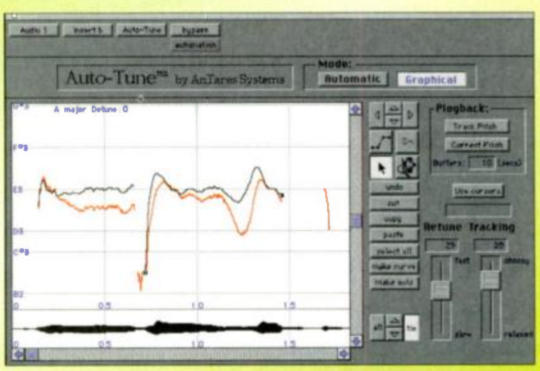



**FIG. A: *Auto-Tune*'s Graphical Mode displays the waveform and pitch profile graphically.**

As a section of the waveform is highlighted, the Make Curve command creates a pitch curve that is based on the original performance. This curve can be globally dragged or reshaped as desired. Graphical mode allows you to maintain the desired amplitude and pitch fluctuations of a performance while adjusting the overall pitch center.

Graphical mode provides much more controlled and detailed pitch editing than Automatic mode, but it is potentially more laborious, and results vary with the skill of the user. Deciding which mode is best for a particular situation is a trial-and-error operation.

"lead" (higher volume, no delay), but the results are often stunning.

## HAPPY HOLIDAYS

While touring with Elvis Costello and Burt Bacharach in the month of October, I still had to complete a holiday CD I'd previously been hired to produce. This meant I was recording during off hours while on the road—not an ideal situation. As with the Warwick project, I had only one shot to record the vocals of three different guest singers. Then, I had only three days to completely mix ten songs in order to meet a mastering and manufacturing deadline.

All three singers gave excellent recorded performances, but under these conditions, just about any vocal track will have a few imperfections. However, recording several takes gave me some flexibility later.

I used the ATR-1 to fix a few spots in a vocal track, allowing me to go for the best performance. In other words, I didn't have to sacrifice a cool vocal passage with a few intonation problems for a more in-tune but less interesting phrase on another take. The ATR-1 let me get more mileage out of the limited amount of material I had, without being bogged down in a time-consuming process and missing the mix deadline. That, my friends, is the sign of a product that is truly productive.

## THE MORAL OF THE STORY

The ATR-1 is one of those products that can save your butt; in fact, it has saved mine twice. It is easy to use and extremely effective once you get a handle on its important parameters. It is very quiet, and it imparts few, if any, perceptible artifacts. Best of all, it can do its thing in real time without destroying the overall feel of the original performance, making it a live-performance tool as well as a studio lifesaver.

Of course, this box isn't perfect; nothing is. I wish it could handle stereo recordings of monophonic instruments and that it had digital I/O and analog level controls. In fact, digital I/O is one of the biggest arguments for the plug-in over the hardware version for serious professional applications.

The ATR-1 is less tweaky than the software plug-in, but it's less complicated to use, too. If a particular track *has* to be absolutely perfect and you have the time to scrutinize and experiment with every little phrase, then you should use the *Auto-Tune* plug-in.

In the end, it was the ATR-1's ease of use that won me over when the heat was on, even though I already had the software version. If my only option for vocal pitch correction had been to use a computer-based plug-in, I probably wouldn't have done anything at all; I simply couldn't afford the additional time it would have taken. As it was, my finished tracks were better because the ATR-1 let me clean up the best vocal performances with very little additional time spent. This is a great trade-off for the more extensive parameter control of the software version. If your need for pitch correction isn't strictly limited to a computer setup, the ATR-1 is the magic black box to have.

---

*Composer and producer* **Rob Shrock** *is the musical director for Burt Bacharach. He has also worked with Elvis Costello, LeAnn Rimes, Luther Vandross, Chrissie Hynde, Stevie Wonder, and a host of others.*

The beat is divided into values ranging from a whole note (one cycle every four beats) to a 32nd note (eight cycles every beat).

Because it's possible to add as many Modulators as you like, you can have an unlimited number of envelopes and LFOs. The more Modulators you add, however, the more complex the Program and the more processing power is required. Even on fast computers, complex modulation Routings may reduce polyphony or increase latency.

Items in the Routings and Modulators lists always appear in the order in which they are created. For example, if you route envelope 1 to control filter 1 resonance, its name becomes Filter 1 Resonance Envelope 1. If, later in the list, you also assign envelope 1 to control volume, its name remains unchanged. It would be more useful if you could rearrange the order so that similar or related Routings and Modulators could be grouped together. This could also identify a Modulator by its most important function rather than its initial destination. For really long lists, it would be even more useful if items could be organized into collapsible groups, like folders in the Mac OS Finder. At the very least, I wish envelopes were grouped with envelopes and LFOs with LFOs. That would make it easier to quickly spot a Modulator in a list.

**Retro AS-1**
**Minimum System Requirements**
**Mac:** PowerPC 120; 32 MB RAM; Mac OS 7.6.1; CD-ROM drive
**PC:** Pentium 200; 32 MB RAM; Windows 95/98; CD-ROM drive; DirectX-compatible sound card

The Editor's Randomize command is a function often found in synth editing software. It randomizes the values of every parameter that is turned on. Many of the resulting timbres are completely useless, and in fact, many are inaudible. Once in a while, though, you may stumble across something that sounds amazing. At the very least, you might find a starting point from which to construct a useful Program.

## EFFECTS AND GLOBAL ISSUES

*Retro AS-1* also includes four effects processors: two insert effects and two global effects. The Insert Effects menu offers Parametric EQ, Shelf EQ, Flange, Chorus, Phaser, Delay, Overdrive, and Distortion. All provide a variety of editable parameters. Delay has two taps with independent delay times and feedback amounts. Maximum delay is just over half a second. Overdrive and Distortion sound quite similar but have different parameters.

The three global effects are Delay, Reflection, and Reverb. This Delay is similar to the insert Delay, except that it provides delay times over a second long and includes a filter to muffle subsequent echoes. Unlike the insert Delay, the two global Delay taps can be independently synchronized to MIDI Clocks. Like LFO sync, the beat is divided into note values. Although it's nothing new, this capability is very cool in any synthesizer.

The Editor's Global page is where you specify a Program's MIDI channel and other basic parameters, such as Volume, transposition, and panning. The amount of polyphony can be limited for each Program. You can also



For manufacturer information, please see Contact Sheet, p. 227

choose from polyphonic triggering and three kinds of legato monophonic triggering. Portamento can glide up or down independently with different portamento times for each direction. The Global page also includes a handy Comments box for typing in notes about a Program file.

## PROCESSING AND MIXING

The *Retro AS-1* MIDI Processor application saves and opens files containing lists of MIDI Setups (see **Fig. 4**). These Setups can be selected with MIDI Program Change commands. Multiple files can be open simultaneously, and you can switch among the first ten windows by holding the Command key and pressing 1 through 10. Each Setup specifies two Programs, each with its own transposition and MIDI channel. These two sounds can be layered, split, or played one at a time.

The MIDI Processor also includes a nifty arpeggiator, which you can apply to either or both sounds in a Setup. It thus allows you to affect both sides of a split or both sounds in a layer. An arpeggio can play whenever more than one key is held down, or it can use a single note as the root of an arpeggiated chord. A List function records and plays back a phrase where each note has the same rhythmic value; notes and Velocities in the list are step entered. In addition, the arpeggiator playback tempo can be controlled manually or synched to MIDI Clock.

Unfortunately, although the MIDI Processor juggles MIDI data, its only output is to the *Retro AS-1* synthesizer engine. As a result, the data that it produces can't be recorded into a sequencer.

The Mixer application provides an onscreen mixing console, which resembles a traditional 16-channel audio mixer. Mixer files contain all the settings for a multitimbral setup. Each MIDI channel has its own channel strip with menus to select Banks and Programs. In addition to Volume faders, there are faders for the two global effects and pan. Mute, Solo, and FX On and Off buttons are also provided, along with a level meter for each channel. As with an audio mixer, the Master section provides faders and meters for the left and right halves of a stereo mix. The Master section is also where you specify the two global effects and their overall levels.

## FINAL BITZ

*Retro AS-1* is a fantastic program; its greatest shortcoming is that many computers aren't fast enough to run it smoothly. The software produces amazing sounds, very much like a real analog synthesizer. The Programs that come with the package have great variety and quality, and concentrate on "synthy" sounds rather than emulative sounds. My hat's off to the programmers who developed this timbral palette; they probably had a great time doing it.

*Retro AS-1* is a lot less expensive than any hardware synthesizer on the market, but only if you've already invested in a speedy computer. To its credit, *Retro AS-1* is probably more powerful than any analog synthesizer around. Where else can you find unlimited envelopes and LFOs? If your computer has the power, it would be a mistake to let this software pass you by without giving it a try. Once you've had a taste, you'll love it.

---

**Geary Yelton** *has been playing with synthesizers for a quarter-century and has been reviewing them for* **EM** *for half that long.*

# DIGITECH

## VOCALIST ACCESS

***An easy-to-use, intelligent harmony processor at an affordable price.***

**By Gino Robair**

The Vocalist Access carries on DigiTech's legacy of "intelligent" harmony processors. In contrast to DigiTech's more expensive and feature-laden Workstation EX and Studio Vocalist EX, the Vocalist Access is an inexpensive and uncomplicated single-channel harmony processor designed primarily for live applications. Up to four harmonies can be used at one time, and they can be individually assigned to the left or right channel.

In addition, the Vocalist Access has a rear-panel aux input that allows access to the unit's reverb while bypassing the vocal processing. And, like the more expensive units, the Vocalist Access uses MIDI messages to change the key or scale, as well as other parameters.

### INSTANT ACCESS

The front panel of the Vocalist Access is spartan in design, with a power switch, a 16-character LCD screen, an XLR mic input, eight small buttons, and three knobs. Six of the buttons, which are physically arranged from low to high, are for choosing the harmony voicing. They include: Bass (octave below the lead), Lower (sixth below the lead), Low (fourth below the lead), Double (adds one detuned copy of the lead), High (third above the lead), and Higher (fifth above the lead). Up to four of these buttons can be selected at once for four simultaneous harmonies.

The four Harmony Control Modes (Scale, Chord, Notes, and Double) partially determine how these various harmonies are used. Presets are selected with either the data-entry knob or a MIDI Program Change message. Of the Vocalist Access's 50 presets, 47 can be edited. DigiTech kindly put the three noneditable ROM presets at the end of the preset list.

Editing the presets is straightforward: press the data-entry knob once and you're in Edit mode, as indicated by the Edit light. Continue pressing the data-entry knob to scroll through the eight editable parameters of each preset. To leave Edit mode at any time, just press and hold the data-entry knob for a moment. This convenient feature frees you from the tiresome process of having to step through every parameter.

Another front-panel knob adjusts the mix levels of the lead voice, harmonies, and reverb. Push the level knob to step between harmony, reverb, and lead, and then turn it to change the level. A series of three green lights tells you which level you're changing, with numerical values indicated on the LCD screen. A small LED next to the level knob, marked MIDI, provides a visual indication that MIDI data is being received by the Vocalist Access.

You can save your parameter settings to RAM using the Store button, and the Bypass button is used to bypass the pitch effects. Next to the XLR input is a handy input-level knob, which controls the line and mic input levels. A small light next to the input-level knob indicates by color the strength of the incoming signal: green indicates that the pitch has been recognized but is below optimal level, orange indicates optimal signal level, and red indicates the onset of clipping.

The rear panel (see **Fig. 1**) is as spare and simple as the front. The unbalanced line input jack prevails over the front-panel mic input when both are used simultaneously; next to the rear-panel input is the ¼-inch aux input. There are MIDI In, Out, and Thru jacks, as well as a footswitch jack that accepts an optional three-button footswitch for remote bypassing, reverb muting, and switching between main and alternate key presets in Scale mode. A pair of unbalanced ¼-inch jacks serve as the main outputs. A button that selects between +4 dBu and -10 dBu serves as the unit's output level control. A variable output control knob on the front panel would have been nice, but because there's only a button, I'm glad it's on the back panel where the setting won't get changed accidentally.

The Vocalist Access has acceptable MIDI implementation. You can control the scale, key, or preset selection using MIDI Note messages. Panning and vibrato depth are modified using Control Change messages.

The user's manual (version 1.1) is designed to get you up and running as quickly as possible, rather than to provide in-depth coverage of the various features. For example, it doesn't identify the kinds of major and minor scales employed; this would have helped in deciding which scale to choose for a particular harmony.

Nonetheless, you can set up the Vocalist Access in minutes, thanks to the intuitive implementation of its various features.

### DOUBLE MODE

The first two presets are devoted to doubling. In this mode, you can add up to four slightly detuned copies of the lead voice for a rich, double-tracked sound. You cannot select harmonies in Double mode; if you want harmonies and doubling simultaneously, you have to switch to either the Chord or the Scale mode, which allow you to select Double as one of your harmony choices. However, in the latter modes, the Double button can add only one extra unison voice.

### CHORD MODE

Chord mode requires MIDI input. The Vocalist Access can recognize 12 chord


**The front panel of the Vocalist Access allows instant access to the six harmony choices, as well as easy and intuitive editing using the data control and level knobs. Having the XLR jack on the front panel is a plus.**

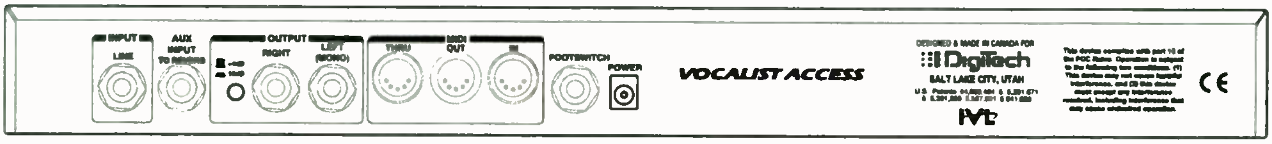



**FIG. 1: The back panel of the Vocalist Access is spare and uncomplicated. The most notable features include the unbalanced ¼-inch line input and aux input. The output level button switches between +4 and -10 dBu.**

types based on incoming MIDI Note messages. The unit then generates harmonies using notes within the input chord, based on the harmony setup you have chosen. For example, if you play an E7 in root position on your MIDI controller, and you have selected a High harmony (a third above the lead), Vocalist Access will place a harmony that is at least a third above the lead voice *within* the E7 chord. That is, if the input note is a nonchord tone (such as an A), the harmony you get will still be an E, G♯, B, or D. Once the harmonization for a particular song is chosen, it can be saved to one of the Chord presets.

The Vocalist Access's chord recognition algorithm is quite sophisticated, allowing you to harmonize on the fly with your controller. Depending on the number of extensions in the chord (sevenths, flatted fifths, and so on), the unit can recognize the chord from as few as two simultaneous notes. An A7(13) chord is recognized by the Vocalist Access as an A6 chord (with no seventh). Play an A7 and then add an F♯ melody note an octave or two above, and the harmonization changes from A7 to A6. Chord types that the unit can read include major, minor, augmented, and diminished chords, with or without added sevenths (see the sidebar "Chord Types").

An interesting parameter in Chord mode is Harmony Hold. When Harmony Hold is on, the Vocalist Access produces harmonies based on the last chord received, even if it receives no new MIDI note messages. In contrast, when Harmony Hold is off, you hear only the original voice when you play a chord. Processing a melody with Harmony Hold on while playing rhythmic block chords produces harmonies with a church organ–style attack similar to the sound of a Mellotron. There's a lot of creative potential with this parameter.

## SCALE MODE

Scale mode creates diatonic harmonies based on the scale chosen. You can select a key and any one of the four major and four minor scales provided. Because the manual doesn't identify the specific major and minor scale characteristics implemented, you'll need to experiment to find the scale with the best harmonization for a particular song. The scale chart at the back of the manual gives some clues about particular scale properties; for example, the "Maj3" scale has a flatted seventh, so the harmonization will have a dominant-chord feel.

The easiest way to determine the best scale to use is to listen to your song with each of them, and decide which one harmonizes the best over the chord changes. As an experiment, I tried "Amazing Grace" with the High harmony above and the Lower harmony below the lead voice. After some trial and error, I chose the "Maj4" scale because it gave me a nice leading tone when going to the dominant chord in the key and the smoothest transitions throughout.

The Vocalist Access lets you switch between the scale types with the data-entry knob for easy comparison. The data-entry knob is quite sensitive, occasionally skipping quickly over settings as I turned it, so that I would accidentally go from "Maj3" to "Min1" and have to dial backward to get to the adjacent "Maj4." DigiTech says this situation has since been remedied.

An added feature of Scale mode is the ScaleSmooth parameter. When ScaleSmooth is on, the harmonies track to pitch inflections in the voice so that bends, vibrato, and scoops are harmonized smoothly without discrete jumps to the next-closest harmony note. If the singer is slightly out of tune, however, the harmonies will be out of tune, as well. With ScaleSmooth off, harmonies will always be in tune with the chosen scale; however, if the singer is off, you risk hearing the harmonies jump back and forth between notes.

You can also select an alternate key for each preset using the optional FS-300 footswitch. Each scale preset can have one alternate setting, which can include a different scale, voicing, and

## Vocalist Access Specifications

| | |
|---|---|
| **Inputs** | (1) balanced XLR; (1) ¼" unbalanced, -10 dBV |
| **Outputs** | (2) ¼" unbalanced |
| **Aux Input (direct to reverb)** | (1) ¼" unbalanced, -10 dBV |
| **Additional Connections** | MIDI In, Out, and Thru; footswitch jack; power jack |
| **Mic Input Level** | -32 to -9 dBV |
| **Line Input Level** | -20 to +6 dBV |
| **Frequency Response (dry)** | 20 Hz–20 kHz (+0/-3 dB) |
| **Presets** | 47 RAM; 3 ROM |
| **Output Level** | +4/-10 dBu switch on rear panel |
| **Signal-to-Noise Ratio** | >92 dB, A weighted |
| **A/D and D/A Converters** | 16-bit sigma-delta |
| **Sampling Rate** | 44.1 kHz |
| **THD + Noise** | <0.04% |
| **Power Supply** | 9 VAC external |
| **Dimensions** | 19" (W) x 1.75" (H) x 5.75" (D) |
| **Weight** | 4 lbs. |

For manufacturer information, please see Contact Sheet, p. 227

key. This is useful when a song changes key in the middle or when you want to change the harmony voicing for the chorus. Not surprisingly, editing the alternate key parameters is quick and painless.

## NOTE MODE

In my opinion, Note mode offers the greatest potential. It takes the note information that it gets through MIDI (single notes or chords), and it creates duplicates of the audio input to complement the note information. For example, if you play a cluster of notes on a keyboard controller, the Vocalist Access will create harmonies that match the notes in the cluster as long as the chord is sustained. The audio input can then move about freely, and the cluster will continue to hold. The held notes follow the formant contours of the audio input, sounding somewhat like a vocoder when used with a human voice.

**The front panel is set up for easy and intuitive editing.**

**DIGITECH**
Vocalist Access harmony processor
$479.95

| | |
|---|---|
| FEATURES | 3.5 |
| EASE OF USE | 5 |
| AUDIO QUALITY | 3 |
| VALUE | 4 |

**PROS:** Easy to use. MIDI controllable. Reverb access via aux input. Nice-sounding reverb.
**CONS:** Timbre of voice is changed during harmony processing. Low voice harmonies are problematic. Harmony algorithm not well suited to use on instruments. No front-panel output control.

CIRCLE #441 ON READER SERVICE CARD

I tried Note mode with my Maestro theremin and got amazing results. A choir of theremins played the chords while the lead theremin moved in and around the choir. The technique worked equally well with vocals.

## LINE INPUT

DigiTech put a line-level input on the Vocalist Access as a way to receive a lead vocal from a mixer: it is not intended to accept an instrument-level signal. Apparently, pitch-shifting algorithms designed for the voice don't work well on instruments. (The exception on this processor is Note mode, which sounded fine.)

I decided to try sending instruments into the line input, just to hear the results. In Scale, Chord, or Double mode, an electric piano sounded best when only one note at a time was played, although there was a slight "chirp" at the beginning of the sound. If any sustain was held over from the previous note, the result was a wild paroxysm of artifacts. Of the instruments I tried through the line input (Gretsch hollow-body electric guitar; Maestro theremin; various sampler voices, such as brass, string, and keyboard sounds), the theremin sounded best.

On the positive side, for those of us who like a bit of uncertainty in our processors, the sounds of various keyboards played through the Scale and Chord modes are fascinating. For example, I chose the fourth major scale in the key of F, added four harmonies, and played in an unrelated key (say, F♯). When I played one note at a time, the timbre had a microtonal sound. Hold down a couple of notes, and it conjures memories of ring modulation. The Vocalist Access treats instrumental sounds in a much different way than DigiTech's ubiquitous Whammy Pedal, and it isn't for the faint of heart. Check out this device if you're looking for wild, analog synth–sounding timbres.

## SOUND COMMITMENT

The unit is called Vocalist Access for a reason: generally, the voice sounds good through it. When you remove the lead voice from the mix, however, you notice that the timbre of the original voice changes during harmonization. In the normal singing register, the added voices have a slight nasal quality, which was enhanced when I switched from a condenser to an inexpensive dynamic mic. When the Bass harmony was applied for harmony an octave lower than the original voice, the new signal sounded unstable. A similar effect occurred with the Bass harmony on a falsetto lead voice. In addition, when I chose a cluster of close harmonies, I had to watch my signal levels carefully because the sound tended to break up and distort when hit with a strong signal.

Ultimately, I found that the Vocalist Access sounded great when the harmony voices were mixed a little behind the lead voice so that they enhanced rather than matched it. The timbre of the harmonies sounded best to my ear when in a register similar to the original voice. Mixing is the key to achieving a natural-sounding blend of harmonies and optimal performance from this device.

## PLENTY OF ROOMS

The Vocalist Access includes a nice balance of room, hall, and plate reverbs, ranging from the long, bright decay of Cathedral to the muffled, short decay of Dark Room. Interestingly, DigiTech has given the Vocalist Access an aux input jack intended for sending line-level signals directly to the reverb section, bypassing any pitch processing. This allows you to use the Vocalist Access on stage as your only processor. If your live setup is as simple as vocals with a guitar or piano, this would work well. The trade-off is that you have to use the same type of reverb on voice and instruments, but in some cases that may not be a problem.

I found that the Vocalist Access's reverb sounded clean, quiet, and unusually responsive, especially on signals sent into the aux input. By carefully balancing the aux level sent to the unit and the internal reverb mix level, it's possible to get a rich sound.

The Vocalist Access allows you to save a particular type of reverb to each preset, which is handy if your set list includes ballads and screamers. On the other hand, if your performance situation requires that you use only one type of reverb in all the presets, you can switch on the Global Reverb parameter to override the preset reverb choices without having to change each one individually. Other than choosing the kind of room and dialing in a mix level, there are no other reverb parameters. Any changes in the reverb setting and level affect the reverb sound for the aux input and the lead and harmony voices.

## HARMONIOUS CONCLUSIONS

The Vocalist Access is an inexpensive way to get an "intelligent" harmony processor and quality reverb into your live-performance rig or personal studio. The harmonies are realistic sounding in the normal singing ranges, the front panel is set up for easy and intuitive selection of harmony voicings, and the editing functions are simple and well laid out.

This processor works best with a MIDI control device, such as a keyboard controller or sequencer; it could, however, be used without MIDI. The ability to route instruments directly into the reverb is an additional bonus, especially when you consider how good the reverb sounds. The Vocalist Access is a pleasure to use and would work well in both amateur and pro settings.

**EM** *Associate Editor* **Gino Robair** *likes MIDI for all the wrong reasons.*

## CHORD TYPES

The Vocalist Access has the ability to recognize chords from a MIDI controller. With as few as two notes, it can recognize major and minor triads, as well as dominant chords. Once the unit has identified the chord, it will harmonize the voice to match, depending on the chosen harmony voicing.

These are the 12 chord types that the Vocalist Access will recognize:

| | | | | |
|---|---|---|---|---|
| major | minor | augmented 7 | diminished | suspended |
| major 7 | minor 7 | | diminished 7 | suspended 7 |
| major 6 | minor 7 (♭5) | | | |
| | minor (maj7) | | | |

ALESIS synthesizers

# Finally, A Sequel That's Actually Better Than The Original

*The QS6.1's four real-time control sliders are assignable to any mod destination, including envelopes, LFOs and even multieffects.*


*With two expansion ports, the QS6.1 can access another 16MB of sounds for a total of 32 meg available at once. Use our QCard™ expansions in your musical style of choice, or burn your own samples to a Flash RAM card using the included Sound Bridge™ software.*

It doesn't usually happen this way. Sequels are supposed to be boring and derivative. But the new QS6.1™ takes the powerful 64 voice synth engine of the original QS6 and supercharges it with double the sound memory, double the expansion capacity, new performance features and much more. So how is it that the QS6.1 got a whole lot better than the keyboard it replaced while actually costing less? The answer is that this sequel is from Alesis – the company that always delivers more than you expect.

QS6.1™
64 VOICE EXPANDABLE SYNTHESIZER

## QS6.1 New Features

- Double the sound ROM of the QS6 (16MB internal)
- Now includes Alesis' stereo grand piano sounds from the QS8
- Enhanced GM sound set
- Double the expansion capacity (up to 32MB total)
- Four control sliders
- Big new LCD display
- New dedicated buttons for Transpose and Sequence Select
- CD-ROM software pack with sequencing, editing, extra sounds, demo programs and more
- Internal power supply
- High speed serial port

**circle #626 on reader service card**

**Alesis Corporation**
1633 26th Street Santa Monica CA 90404 800-5-ALESIS alecorp@alesis1.usa.com www.alesis.com
For more information on the new QS6.1, contact your Authorized Alesis Retailer or visit our web site.


ALESIS

**SEE US AT NAMM BOOTH #1033**

# STEINBERG

## WAVELAB 2.0 (WIN)

***Record, edit, master, and burn your own CD with a single piece of software.***

**By Scott R. Garrigus**

Steinberg's *WaveLab* digital audio editor for Windows has been around for a few years, but with the introduction of version 2.0 it has clearly evolved into a robust and versatile stereo audio editing package. In fact, it now includes most everything you need to record and edit audio, as well as to produce and master CDs with professional results.

Aside from the usual tools found in most audio editing software, *WaveLab* boasts a number of high-end features, including support for 32-bit, 96 kHz I/O hardware; real-time effects processing (which works on prerecorded files or live input); file analysis tools; a built-in audio database; dedicated sampler support with looping functions; and CD-burning capabilities. With such a powerful and feature-laden package, this program can tackle almost any audio-editing task.

### EASY GOING

If you're comfortable working with the Windows 95/98 interface, you'll be equally comfortable using *WaveLab.* Toolbars representing all the major functions in the program can float freely or dock along any boundary of the main window. Furthermore, you can have as many audio files open as you like (disk space permitting), each with its own window (see **Fig. 1**).

For each open window, *WaveLab* provides a Snapshot function, so you can store and recall up to eight zoom and position settings for quick and easy navigation. *WaveLab* also allows you to save the overall layout of your work environment with its Screen Layouts feature. You can store any number of Screen Layouts and even assign a keyboard equivalent to each one for rapid recall. Unfortunately, the Screen Layouts function doesn't encompass every aspect of the work environment. For instance, it doesn't affect the positions or visibility of the toolbar palettes.

Going beyond the standard interface conventions, *WaveLab* provides "nonmodal" dialog boxes. This means that, if you're applying EQ to a file, for instance, you can leave the EQ dialog box open while switching back to the waveform display for further editing.

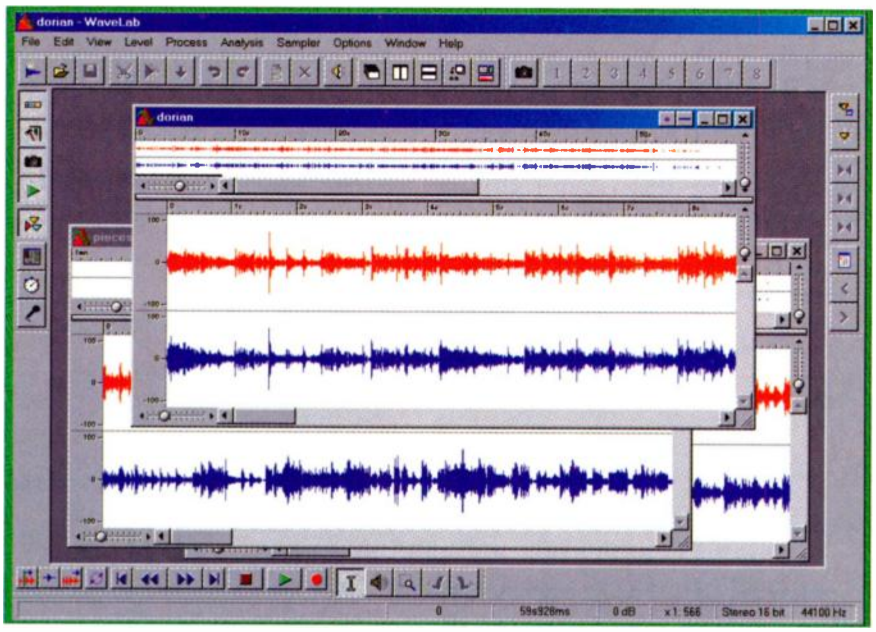



**FIG. 1: *WaveLab* provides a separate window for each open audio file. Each window has two panes: the Overview pane for navigation, and the Editing pane, in which all audio manipulation takes place.**

You can even switch to another window to work on a different file. In fact, you can access any of the program's functions while a dialog box is open, and you can have multiple dialog boxes open simultaneously. This feature alone can save you lots of time.

*WaveLab* also provides an unusually high degree of customization. Through its use of context-sensitive pop-up menus, the program lets you change just about every aspect of a waveform window: Right-click on the horizontal ruler to display elapsed time, samples, time code, meter, or file size. Right-click on the vertical ruler to display audio level as a percentage, in decibels, or in a 16-bit decimal range. You can change the style of both rulers, from the font to the color of the ruler markings shown. Right-click on the main editing area of the window to specify which parts of the window are shown (such as rulers and markers); you can also control the appearance of the axis lines and colors for every part of the display.

To help streamline editing tasks, *WaveLab* lets you store presets not only for its effects but for nearly all its processing features. These features include creating crossfades, loops, and even batch processes. The built-in batch processor allows you to run audio files through any of *WaveLab*'s processing functions.

With presets, you can practically automate all of your audio processing chores. One minor complaint, however, is that it's impossible to assign keyboard equivalents to presets. Granted, there are too few buttons on the computer keyboard for every preset you might want to create, and *WaveLab* does let you name your presets and get to them with only a couple of mouse clicks. Still, calling up some favorites with a simple keystroke would be nice.

### RED LIGHT, GREEN LIGHT

Clicking on the Record button in *WaveLab* opens a dialog box in which you choose a file format and set your input levels. The dialog box also provides a meter display with left and right peak indicators and a readout showing the amount of time recorded and the amount remaining (based on disk capacity). Depending on your sound card, clicking on the Mixer button expands the dialog box to reveal faders for all

the available inputs. For example, on a Sound Blaster card, controls for the Synth Input, CD Input, Line Input, and Mic Input are displayed (see **Fig. 2**).

When first installed, however, *WaveLab* defaults to the Microsoft Sound Mapper for its sound-card playback and recording settings. This makes the Mixer button inaccessible. To make the Mixer button work, you need to change the card settings in the program's Options/Preferences menu to correspond with your hardware.

*WaveLab* can record in a variety of file formats with sampling rates ranging from 2 to 96 kHz and resolutions ranging from 8- to 32-bit. The options you can use, of course, depend on the capabilities of your sound card. In addition to the usual mono and stereo formats, *WaveLab* supports a dual-mono format for working with programs (such as Digidesign's *Session*) that handle stereo files as two mono files. *WaveLab*'s dual-mono feature allows you to open two mono files and edit and process them as though they were a single stereo recording.

Unfortunately, *WaveLab* supports a smaller range of file formats than some of its competitors do: the program handles only WAV, AIFF, AU, RAW, and Ensoniq Paris formats. Multimedia audio producers might find this rather limiting. The program does, however, let you save (but not import) files to any compressed format supported by the audio codecs that you have loaded in your system. (Windows comes with a number of audio compression drivers that allow you to save your audio to compressed formats, such as ADPCM.)

Once you've recorded a file, you can play it back in several ways. For example, you can start playback either from the beginning of the file, from the current cursor position, or from the start or end of a selection; you can also set the playback characteristics for the program's loop mode (say, loop the entire file or loop a selection). Moreover, *WaveLab* provides special "skip" modes that allow you to jump past selected regions during playback. The program also has a full arsenal of marker functions that let you create markers at the current cursor position and initiate playback from any marker using the Markers window.


**FIG. 3: The Master Section contains the controls for *WaveLab*'s real-time effects-processing functions. Here we see the EQ, Chorus, and Reverb effects in use, with the remaining three effects "slots" still free.**

Regrettably, synchronization is one significant recording/playback feature that *WaveLab* does not provide. The program can neither transmit nor receive SMPTE time code (or MIDI Time Code) for synchronizing to an external source. Because of this, *WaveLab* is unsuitable for many audio-for-video recording situations.

## IT SLICES, IT DICES

Editing in *WaveLab* is as intuitive as the other functions in the program. Aside from the usual Copy, Cut, and Paste commands, the program has other special pasting commands, including Append (attaches to the end of the file), Prepend (inserts at the beginning), Paste Multiple Copies, and Mix. You can also manipulate the audio with the program's extensive drag-and-drop support.

Creating audio loops is a snap. First, click on the Options menu and make sure that the Snap to Zero-Crossing function is activated. Next, drag the mouse to select the portion of audio that you want to loop. The Snap to Zero-Crossing function automatically extends or retracts the selection end points to the nearest zero-crossing. If you want to use the loop as a separate audio file, simply click and drag the selection to an empty area on the *WaveLab* work space. A new window opens with your audio sample inside. You can also define the selection as a loop within the original audio file by using the Create Loop button in the Markers toolbar. This automatically places start and end loop markers at the beginning and end of your selection.

If you make a mistake, it's easily reversed with a click of the Undo button. *WaveLab* provides an unlimited (based on disk space) Undo/Redo function. Each open audio file window has its own separate undo "history," but because there's no history display, you can't redo from a specific point. You

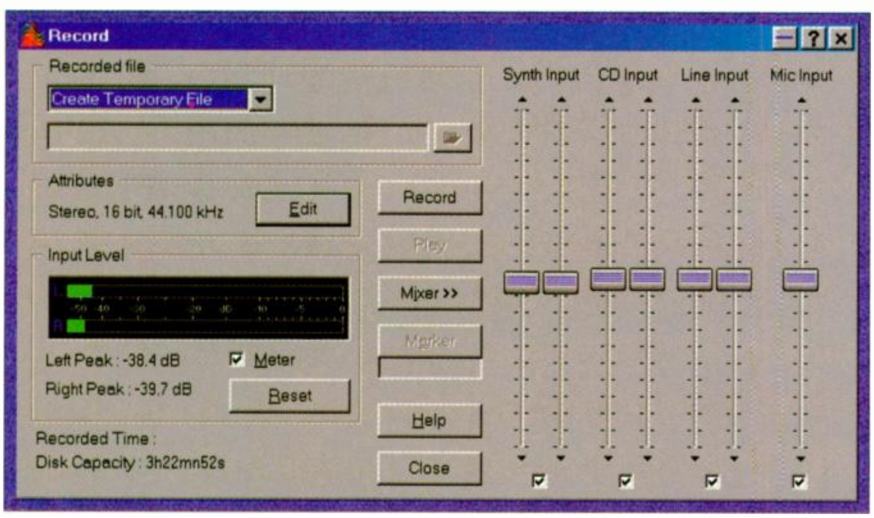



**FIG. 2: *WaveLab*'s Record dialog box includes audio level meters, peak indicators, and onscreen fader controls for setting levels.**

can get around this, somewhat, by selecting an entire file and dragging it to a new window. Then you can minimize the file and continue working on the original. Nonetheless, a history display would be much more useful.

## TRANSFER POINTS

Sound designers and those who work with samplers will especially appreciate *WaveLab*'s dedicated sampler support. You can transfer audio samples to and from any sampler that supports the generic SDS or generic SMDI communication protocols. *WaveLab* also offers support for specific samplers, including models from Akai, Ensoniq, E-mu, Kurzweil, and Roland. I used an Akai S3200 to test the program. Everything worked fine, except that I was forced to use the excruciatingly slow SDS protocol. (Akai uses its own proprietary data-transfer protocol for the S3200, rather than the standard SCSI/Windows protocol.)

Before transferring your loops, you can fine-tune them with *WaveLab*'s Crossfade Looper and Wave Equalizer. The Crossfade Looper allows you to see and precisely manipulate the splice between the start and end loop points. You can manually nudge the loop points or have the program automatically find the best loop points for you. When you think you've found a good point, you can store it—and up to four more candidates—in the five available buffers. You can then easily compare them to find the perfect choice.

**Creating audio loops is a snap.**

*WaveLab* also enables you to create crossfades at both the start and end points for smoother loops. If you have an unusually stubborn piece of audio, the Wave Equalizer can help to even out changes in level and timbre by cutting the loop into slices and mixing them back together into one sound. You have to actually hear this feature to appreciate it.

## DESTRUCTIVE BEHAVIOR

*WaveLab*'s destructive processing commands are not, in themselves, particularly unusual. The standard options are available, including Normalize, Change Gain, Invert Phase, and Eliminate DC Offset. The Dynamics command provides a nice graphic interface for compressor, expander, noise-gate, and limiter effects. There are also commands for Time Stretching, Pitch Correction, Chorus, and EQ. What makes *WaveLab* stand out in this area is that all these processes can be performed while the current audio file plays in the background.

For instance, if you're working on a sample loop and you have the program set for continual play, you don't have to stop playback to process the audio. If you want to add some EQ, just click on the Process/EQ menu, input your settings, and hit the Process button. As soon as *WaveLab* finishes processing, you hear the changes in the looped playback. If you don't like the change, click the Undo button to instantly restore the original. This functionality, combined with the program's nonmodal dialog boxes, makes for a truly productive environment.

## REAL-TIME EFFECTS

During playback, all of the audio passes through *WaveLab*'s Master Section


**FIG. 4: Each VST effects plug-in presents an effects-rack appearance similar to its hardware equivalent.**

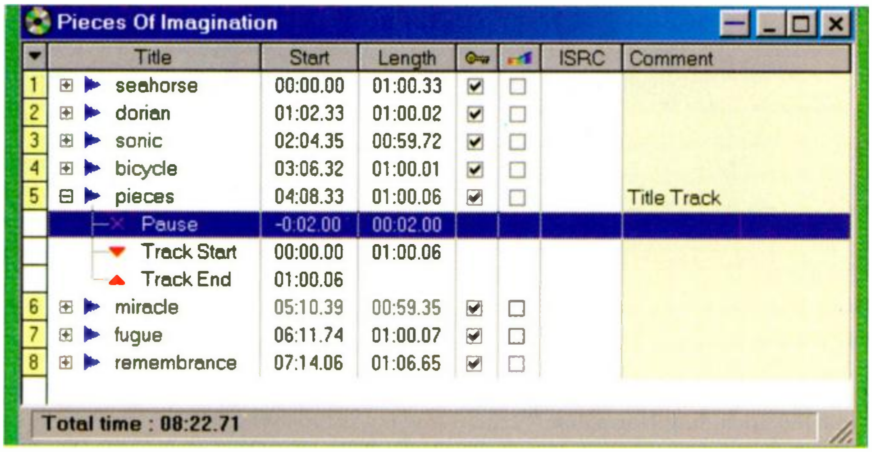



**FIG. 5: Before burning a CD, you must assemble a track list, as shown here. Clicking on the plus sign next to an entry reveals the pause length.**

before being routed to your sound card's output. The Master Section incorporates *WaveLab*'s real-time effects plug-in architecture. Following an "effects rack" metaphor, it lets you connect up to six effects modules (depending on your computer's processing power) in series by simply selecting one of the available drop-down menus and choosing an effect (see **Fig. 3**).

*WaveLab* supports Steinberg's VST plug-in format, in addition to Microsoft's DirectX, so you can easily expand your processing arsenal as needed. Even so, the program ships with 13 of its own plug-ins, including an AutoPanner, Chorus, EQ, Echo, Reverb, and Puncher (a kind of spectral enhancer that adds harmonic content to accentuate peaks and make a "punchier" sound without causing clipping).

With its extensive plug-in support, *WaveLab* can quickly accumulate a large collection of effects, so the program also includes a handy Plug-in Manager that enables you to organize and group plug-ins and determine how they appear in menus.

*WaveLab*'s VST plug-ins have a graphic interface designed to look like their hardware-based equivalents, complete with onscreen buttons and parameter dials (see **Fig. 4**). Although these onscreen controls can be cumbersome at times, a simple double-click allows you to change any parameter by typing in a value. Of course, you can also save presets for your favorite settings.

The only effect in the group that I felt wasn't up to snuff was the Reverb module. Instead of providing a nice smooth effect, the Reverb plug-in added a slight metallic edge to the sound, such as you might get from an inexpensive rack-mount box. No matter how much I tweaked the effect, the edge remained.

You can apply Master Section effects to an audio file destructively, using the Apply button, or you can initiate playback and hear the effects applied in real time. Most processing is done internally with 64-bit floating-point precision (some tasks involve 32-bit floating-point processing), so the output is crisp and clean. You can compare dry and processed signals with the Global Bypass button or with the individual bypass buttons available on each effect. You can also save Master Section presets, which store all the settings for each processor in use, along with dithering options and the window position of each plug-in. That's a very nice touch.

## GET ORGANIZED

If you typically have a hard time keeping your project files organized, you'll appreciate *WaveLab*'s Database and Project features. A Database allows you to organize all of your audio files, whether they're stored on a hard disk, removable disk, CD-ROM, or floppy disk. You can assign categories and keywords as well as comments to each file and then search for specific criteria to find the files that you need. If you select a file located on a removable medium that is currently not loaded, the program prompts you to insert the correct disk or CD-ROM.

> **WaveLab 2.0**
> **Minimum System Requirements**
> Pentium 166; 16 MB RAM;
> Windows 95/NT 4.0

You can add files to a Database manually, or you can have the program scan various media for files that match your name, file format, file size, and date criteria. Files that match the settings are automatically added to the Database. Unfortunately, because *WaveLab* supports a select group of common file formats (WAV, AIFF, AU), those are the only file types you can include in a Database; that fact keeps the program from being a truly universal file organizer.

A Project allows you to organize your files but only for a single session (unlike a Database, which is a more global organizational tool). You can also group your files into folders, so you can have, for example, all your drum loops in one folder and your bass loops in another. You can add files to a Project by dragging and dropping them from a Database or by selecting them from a disk using a file selector dialog box. Either way, files added become part of the current Project, which stores all the window settings for each file, including their locations, size, and zoom settings. This is a great help when you need to take a break and then return to a Project and pick up where you left off.

## FEEL THE BURN

One of *WaveLab*'s main attractions is its inclusion of CD-burning capability. With some similar products, you have to purchase a separate piece of software to acquire this feature, so it's nice to have it included as part of the package. The program supports only disc-at-once mode and won't allow you to store files on a CD in data format. In general, however, this lack of options makes the program easier to use: I was able to burn a disc on my first try without any problems.

The CD function allows you to assemble a group of files as a track list, or you can use a single audio file that is segmented with markers as the source of the CD tracks. Information shown for each track in the list includes the track title, start time, length, copy protection status, emphasis status, ISRC (International Standard Recording Code), and comments (see **Fig. 5**). All these parameters can easily be edited, along with the pause length between tracks. You can listen to how your finished CD will sound by double-clicking on the first track in the list. The program will then play through the list, adding the proper pause lengths.

For preparing CDs, *WaveLab* includes its unique Meta-Normalizer tool. The Meta-Normalizer analyzes your files as a batch process before burning your CD. It then normalizes your files (based on algorithms and analysis) so that each track has the same *subjective* volume. *WaveLab* also provides tools for analyzing and comparing files and for detecting and reporting data errors, which can save you much grief if you're replicating a large number of discs.

Writing speed is adjustable from 1× to 6× and, of course, depends on your hardware. Before committing to actual media, you can do a test burn to make sure the CD can be written at the chosen speed. And if you plan to have your disc professionally duplicated, you'll appreciate the added bonus of being able to print cue sheets based on your customized templates and the current track list.

### LEADER OF THE PACK

*WaveLab* uses a copy protection scheme that can be annoying: you must periodically insert the original CD before running the program. (I must confess, however, that I prefer this scheme to the dreaded dongle.) Its lack of synchronization and support for only a few common audio file formats may also make it less than appealing to some users in the multimedia or film/video post-production fields. On the other hand, sound designers, sampler owners, and musicians interested in mastering audio to CD shouldn't think twice about buying this excellent software package.

*WaveLab* is easy to use and provides all the necessary tools to get you from recording to CD mastering. Extras such as its multitasking capabilities and real-time effects architecture help reduce project time and provide an intuitive and creative work environment. Furthermore, its high-end audio support, with 32-/64-bit internal processing, and its support for file resolutions as high as 32-bit put this program at the head of the pack. And with its good documentation and online help, *WaveLab* is well worth the price. ❁

**STEINBERG**
WaveLab 2.0
audio editor
$499

FEATURES ■■■■
EASE OF USE ■■■■
DOCUMENTATION ■■■■
VALUE ■■■■
1 2 3 4 5

**PROS:** Unlimited undo/redo. Direct sampler support. Built-in CD burning and batch processing. High-end hardware support. Has 32-bit internal real-time effects processing. Multitasking capabilities. Easy operation. VST and DirectX effects plug-in support.

**CONS:** Reverb effect needs improvement. No random access to undo/redo history. No support for SMPTE or MIDI Time Code synchronization.

**CIRCLE #442 ON READER SERVICE CARD**

# ROLAND

## VS-840

### *The VS-880's little sibling measures up.*

By Bob O'Donnell

With the introduction of the VS-880 several years back, Roland practically redefined the portable digital studio. True, modular hard-disk recorders with onboard mixers (digital and analog) had existed for some time, and some companies—including Roland, with the DM-800—had released units with hardware mixing surfaces. Nevertheless, the VS series raised the bar for such products by providing a full-featured mixing surface with internal and MIDI automation and by going all-digital. Later, CD-burning capabilities and software links were added. Not surprisingly, the idea caught on and made the VS-880 one of the company's greatest successes.

But for all the great features the VS-880 offers, its user-interface is complex and, well, let's just say the word "intuitive" doesn't exactly spring to mind. To address that issue, as well as offer a less expensive version of the technology, Roland's product designers came up with the VS-840, an 8-track portable digital studio that, in some ways, surpasses the capabilities of its more expensive predecessor.

## THE MIXER

The first four channels of the VS-840's 8-channel digital mixer are mono, and channels 5/6 and 7/8 are stereo pairs. As with most such devices, the unit has faders and familiar tape deck–style "transport" controls, including a handy Return to Zero button. It also offers a variety of locate and mark points for jumping to different parts of a song, setting in and out points for punch-in recording, and so on.

A set of dedicated cue controls allows you to adjust each track's playback levels independent of the mixer's input faders. With this feature, you can create a separate cue mix while overdubbing, for example. I also appreciate the inclusion of Solo buttons on each channel.

The 3-band channel EQ comprises semiparametric low and high bands and a fully parametric mid band. You can use it while recording, bouncing, or mixing. One neat touch is that the graphic display will show you the EQ curve you create as you make adjustments. This EQ is separate from, and can be used in addition to, the onboard digital effects.

To access EQ functions and manually set input channels, panning, internal effects, and aux-send levels, you use the multifunction Channel Parameter buttons, which are located above the channel faders. One undocumented trick is that, if you hold down the Shift button before you hold down a Channel Parameter button, you can temporarily assign the Track Cue knobs to make adjustments to the selected channel parameter across all the mixer channels. So, for example, if you press and hold Shift and then select Pan, you can use the Track Cue knobs to make quick adjustments to all channel pan settings. Because you have to keep holding down the Shift key, it's a bit clumsy, but it works.

## INS AND OUTS

The rear panel sports four balanced ¼-inch mic/line inputs. An additional high-impedance guitar input feeds channel 1 (see **Fig. 1**), which is a nice touch. (The guitar input is defeated when you use the mic/line input.) Each of the four inputs features a sensitivity control (with a peak LED) that lets you plug in anything from dynamic microphones to line-level synth signals to +4 dBm pro-level gear. The unit also offers two RCA inputs that can be configured as a stereo effects return or as additional line inputs that feed channels 3 and 4; these are disabled if you use the regular ¼-inch channel inputs.

You get a stereo master output pair and a stereo monitor/auxiliary output pair, all on RCAs. You can also use the monitor/aux outputs as two independent mono aux sends. In addition, the VS-840 features a headphone jack and both coaxial and optical S/PDIF outputs, which allow you to mix down straight to a DAT or other digital mastering device. Other connections include a footswitch jack and MIDI In and Out jacks.


**Roland's VS-840 8-track portable digital studio is designed for the recording musician on a budget who wants better sound quality and more features than multitrack cassette decks or MiniDisc systems provide.**

## MAKING TRACKS

The VS-840 offers up to 64 virtual tracks (8 per physical track), which you can use to record alternative takes, submix tracks, etc. All recordings are stored on the device's built-in Iomega Zip removable-cartridge drive, which can hold 100 MB of data on relatively inexpensive media. You can record up to four tracks at once and play back all eight tracks simultaneously.

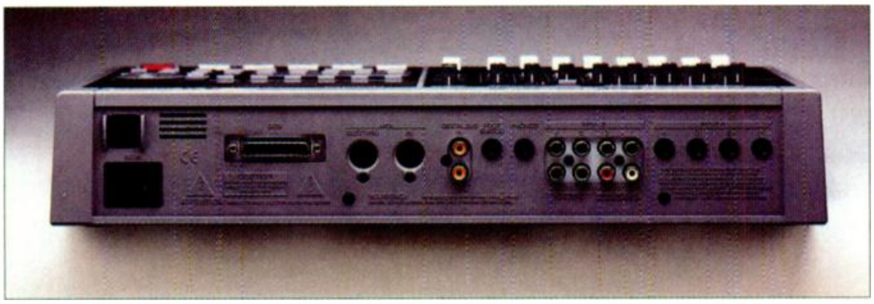


**FIG. 1: The VS-840 includes a dedicated guitar input, as well as other analog inputs, and both analog and digital outputs. Shown here is the VS-840S, which includes a SCSI port that allows you to back up data from the internal Zip drive.**

Recordings can be made using one of four recording modes, which employ different types and amounts of audio compression as well as different sampling rates (44.1 kHz or 32 kHz). Unlike the VS-880 and VS-1680, the VS-840 cannot record uncompressed audio.

To my ears, the VS-840's default recording mode (which is a notch below its best possible mode) sounds quite good. I heard no distinguishable artifacts, though there are undoubtedly subtle losses due to compression. The EQ generally sounds good, as well, although it doesn't sound as "round" on the low end as, for instance, the Fostex FD-4's analog EQ does.

One other neat sound-related capability of the VS-840 is that it supports variable-pitch playback for correcting a slightly out-of-tune prerecorded piece or for creative effects. The machine does not, however, do time compression or expansion.

The total recording time on a single 100 MB disk ranges from 37 track minutes at the highest quality to 103 track minutes at the lowest. Dividing that recording time by eight tracks means that you could have as little as 4.5 minutes of total song time if all eight tracks were completely filled with audio data. Because most tracks have gaps, however, and because silence doesn't take up space, even the highest-quality setting should let you comfortably record most song-oriented projects.

An optional SCSI port costs $150 plus installation charges, or you can buy the VS-840S with the SCSI board preinstalled for an extra $200. However, in the first version of the VS-840 firmware, the SCSI port can only be used to copy data from the internal Zip drive to a single, external Zip drive, which is useful but not adequate for those who wish to record extended works at the highest quality. Let's hope that a later revision will add the ability to record directly to other external SCSI devices with larger capacities.

## DIGITAL EFFECTS

The VS-840 features one stereo effects processing chip, whereas the optional VS8F-1 Effects Expansion board for the VS-880 (included with the newer VS-880EX) has two DSPs. The 26 effects algorithms in the VS-840 are all taken from the VS8F-1, which offers 30 algorithms. You get plenty of editable parameters in most of the effects. In fact, one of the "Guitar Multi" algorithms offers over 60 parameters and seven different simultaneous effects for every patch that uses it.

The effects range from a wide variety of COSM physically-modeled guitar effects—such as distortion and other guitar-oriented patches—to reverbs, delays, vocoder, and even Roland's RSS 3-D effects (see the sidebar "The Effects List"). The VS-840 doesn't have mic-simulation programs, as the VS-880 does (version 2.0 and later), which is unfortunate because those effects seem particularly well-suited to budget studios.

The 300 preset effects sound very good, and there's room for a bank of 100 user-defined effects patches. However, the COSM-based guitar-amp simulations are surprisingly noisy. They're usable, but I was disappointed by the background hiss that popped in every time I selected a guitar effect. According to Roland representatives, the company is aware of the problem and is working on a new set of quieter presets. In the meantime, you can reduce the noise levels on many of the presets by adjusting their parameters.

## THE RIGHT STUFF

One of the biggest improvements the VS-840 offers over the VS-880 and 880EX is a graphical, icon-based user interface that takes excellent advantage

### VS-840 Features

| | |
|---|---|
| **Tracks** | 8 |
| **Virtual Tracks** | 64 (8 per track) |
| **Simultaneous Record/Playback Tracks** | 4/8 |
| **Mixer Channels** | 8: (4) mono, (2) stereo pairs |
| **Faders** | (6) channel, (1) master; 60 mm |
| **EQ** | 3-band: semiparametric low and high bands, parametric mid band |
| **Effects** | 26 algorithms, 300 presets, 100 user programs |
| **Locate Points/Mark Points** | 8/1,000 |
| **Mixer Scenes** | 8 |
| **Internal Disk Drive** | Iomega Zip 100 MB removable cartridge |
| **Internal Memory** | 200 Songs (per disk) |
| **Channel Inputs** | (4) ¼" unbalanced mic/line; (1) ¼" hi-Z guitar (ch. 1); (2) RCA line (ch. 3 & 4) |
| **Master Mix Outputs/Channel Outputs/ Headphone Out** | (2) RCA/0/(1) ¼" stereo |
| **Aux Sends/Channel Inserts** | (2) RCA (double as monitor outs)/0 |
| **Digital Inputs/Outputs** | 0/2: (1) stereo S/PDIF optical, (1) stereo S/PDIF coax (both carry master mix) |
| **Additional Connections** | MIDI In and Out; SCSI DB25 (VS-840S only); ¼" footswitch |
| **Display** | 69 x 25 mm backlit LCD |

For manufacturer information, please see Contact Sheet, p. 227.

of the unit's large, backlit LCD. Even with the breadth of functions offered, the interface's operation is intuitive, and the features are logically laid out. On a device as sophisticated as this one, I think that's critically important, and it's one of the main reasons I believe this unit stands out over its competitors. If you've become frustrated by the confusing interfaces found on many of today's electronic music devices, I urge you to check out the VS-840, if for no other reason than to see how an interface can be done right.

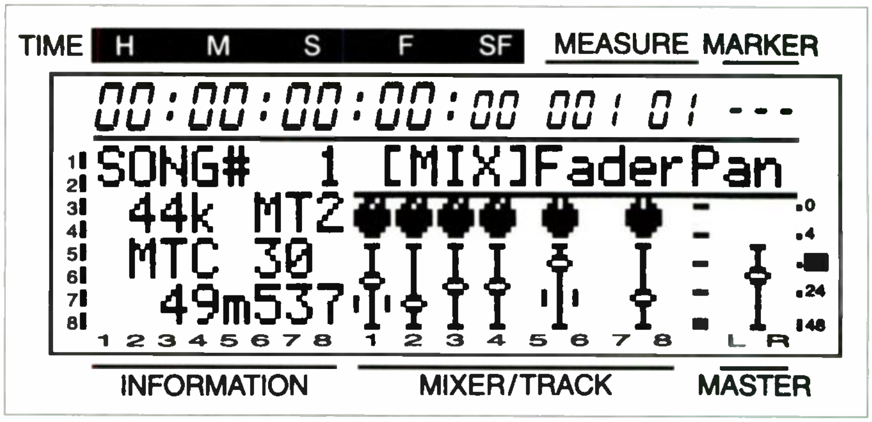
On top of that, I was pleasantly surprised to find that the VS-840's documentation is thorough, well written, and informative—which isn't typical of Roland manuals.

One of the best features of the interface is EZ Routing, which walks you through the often-confusing process of assigning input signals to mixer channels and tracks. You can use EZ Routing for recording, bouncing tracks, and mixing tracks. When you first hit the EZ Routing button, you navigate through several pages of information where you assign inputs to mixer channels and tracks (real or virtual). Along the way you can set external effects-send levels, choose to record or just monitor the track with internal effects, and set panning levels for each channel. Best of all, the unit is smart about what tracks and channels are available, so you can usually just answer yes or no to the questions it asks and it will set you up with the routing you need.

If you want to, you can skip the EZ Routing feature and make any assignments on your own with User Routing, but EZ Routing is so quick that I imagine most people will use it all the time. (In addition, it ensures that you remember to set parameters you might otherwise forget.)

## PARAMETERS ON DISPLAY

The VS-840's LCD is capable of displaying a wide variety of information (see **Fig. 2**). In addition to the parameter pages, the display includes level meters for the channels and the master bus. A Playlist view shows you which tracks have audio on them. You can also graphically view the fader, pan, and track-cue settings of each mixer channel. You can have the displayed faders match where the actual faders are, or, if you call up a Scene with different levels, you can show where that Scene's fader settings are and ignore the actual faders' current positions.




COURTESY ROLAND CORPORATION U S

**FIG. 2: The VS-840's backlit graphic display shows off a wide variety of information, such as this screen, which displays the current fader and pan levels, as well as the overall recording and synchronization settings.**

You can also use the graphic display to perform any of the editing functions you'd expect to find on a digital recorder, such as cut, copy, paste, move, and insert. You can perform these operations on single tracks or on sets of tracks. The Scrub function, which also provides a simple visual waveform display, is useful for setting the proper edit points.

## GETTING IN SYNC

In addition to working well as a standalone device, the VS-840 has a number of features for use in conjunction with other MIDI-based devices, such as sequencers and drum machines. To begin with, it can send (but not

### THE EFFECTS LIST

The VS-840 has one built-in stereo effects processor, which uses any of 26 algorithms. Each algorithm is composed of some combination of the following effects: acoustic guitar simulator, chorus, compressor/limiter, de-esser, delay, enhancer, 3- or 4-band EQ (high and low shelving, 2-band parametric mid), stereo flanger, Lo-Fi (includes AM radio simulator, gramophone record simulator, and low-fi processor), rotary speaker simulator, noise suppressor, overdrive, stereo phaser, pitch shifter, preamp, reverb, ring modulator, RSS (2-channel, 3-D processor), RSS (panner), space chorus, speaker simulator, tremolo/pan, vocoder, and wah. The unit comes with a total of 300 effects presets.

receive) MIDI Time Code and MIDI Clock and can send and receive MIDI Machine Control. It maintains an internal tempo map that you can use to generate a metronome click while recording, and it can display your song location in bars and beats instead of just absolute time. This tempo map is also used to generate MIDI Clock and Song Position Pointer messages.

The tempo map supports multiple tempo changes, but you have to create a tempo map for each new tempo. Unfortunately, the VS-840 does not support Tap Tempo—at least, not in this version of the operating system—despite the confusingly named Enter/Tap button. For guitarists and others who may prefer recording a guitar part first and then tapping in the tempo to create a tempo map, that's an unfortunate omission.

One other great MIDI feature that deserves mention is the fact that you can upgrade the VS-840's operating system via SysEx. As with other VS products, Roland distributes OS upgrades embedded in MIDI sequences that you can download for free from its Web site, load in a sequencer, and play into the VS-840's MIDI In port. I was able to use this to upgrade from version 1.00 to 1.02 (which has some minor bug fixes) during the course of the review. This is a very clever idea I'd like to see other manufacturers imitate.

### ON THE SCENE

Despite all these powerful features, the VS-840 lacks dynamic mixer automation, which is a bit of a disappointment. If you want to automate real-time changes, you need to step up to the VS-880 or VS-1680.

However, the VS-840 can store up to eight mixer scenes per song. Each scene can include completely different level, pan, EQ, and effects settings, which lets you try out different mixes. Unfortunately, the machine can't show you which scene you're currently using, so you have to be conscious of what scene you're selecting.

### THE NEXT GENERATION

It's clear from working with the VS-840 that this is a second-generation product that benefits from all the feedback that VS-880 users have given to Roland. The company smartly included the effects processor as standard equipment, added a dedicated guitar input, greatly improved the graphic LCD, and improved the overall usability of the machine. At the same time, Roland also had to omit certain capabilities to meet the goal of bringing the unit to market at half the original VS-880's price. The result doesn't feel terribly compromised, though, and a few software upgrades

**ROLAND**
VS-840 portable digital studio
$1,399
VS-840S (with SCSI)
$1,599

| | |
|---|---|
| FEATURES | 4 |
| EASE OF USE | 5 |
| AUDIO QUALITY | 4 |
| VALUE | 4.5 |

1 2 3 4 5

**PROS:** High-quality sound. Affordable. Good built-in effects. Excellent graphic interface. Great documentation.

**CONS:** Recording time limited at highest quality rates. Can't record without compression. Lacks dynamic mixer automation. Guitar effects are a bit noisy. No digital inputs.

CIRCLE #443 ON READER SERVICE CARD

## VS-840 Specifications

| | |
|---|---|
| A/D Converters | 20-bit, 64x oversampling |
| D/A Converters | 20-bit, 128x oversampling |
| Sampling Rates (kHz) | 44.1, 32 |
| Sampling Resolution | 16-bit |
| Internal Processing | 24-bit (mixer section) |
| Frequency Response (@ 44.1 kHz) | 20 Hz–21 kHz (+1/-1.5 dB) |
| Total Harmonic Distortion (@ 44.1 kHz) | <0.08% in best record mode (MT1 mode; input sensitivity -10 dBm, 1 kHz at nominal output level) |
| Residual Noise | <-91 dBm (input terminated with 1 kΩ, input sensitivity +4 dBm, typical) |
| Dynamic Range | 96 dB |
| Nominal Input Level | -50 to +4 dBm |
| Nominal Output Level | -10 dBm |
| Dimensions | 16.188" (W) x 12.125" (D) x 3.5" (H) |
| Weight | VS-840: 9.94 lbs.; VS-840S: 10.38 lbs. |

could make this an absolutely killer home-studio machine.

In particular, I'd like to see recording time increased by allowing you to use the optional SCSI port for recording, support for dynamic mixer automation and Tap Tempo, more detailed waveform editing, a few more effects choices, quieter guitar effects, and maybe even a built-in tuner. (Company representatives strongly hinted that version 2.0 software for the VS-840 was set for a January 1999 NAMM debut, so it should be available by the time you read this. Check Roland's Web site for information.)

That said, the current version of the VS-840 is an excellent choice for recording musicians on a budget, regardless of their level of experience and whether or not they use computers and other MIDI gear. It costs a little bit more than some of its competitors, but its features and, most importantly, its great user interface are well worth the difference in cost. Overall, it just feels better and more like a serious piece of recording gear.

Professionals and experienced personal studio owners who have the money will probably be better served by the VS-1680 or VS-880EX, with their advanced features and support for uncompressed recording. But the VS-840 is bound to become the centerpiece of some great budget studios.

---

**Bob O'Donnell**, *former editor of* **EM**, *hosts the* O'Donnell on Computers *radio show, which you can hear via RealAudio on the Web at www.everythingcomputers.com. He also writes a weekly column for* InfoWorld Electric, *the online version of* InfoWorld *magazine, at www.infoworld.com.*

# CYCLING '74

## M 2.5 (MAC)

***A new company reintroduces an old favorite at a bargain price.***

**By John Duesenberry**

The end of the 1980s was a period of creative ferment in the field of algorithmic and interactive composition software. *Sound Globs, Tunesmith, Ovaltune, Upbeat, Algorithmic Composer, Jam Factory*, and *M*—to name a few worthy programs—hit the marketplace almost simultaneously. Perhaps the music world wasn't ready for them, though: many of these products, alas, disappeared nearly as quickly as they were introduced.

My personal favorite from that era was *M*, brainchild of composer Joel Chadabe and three of his students, Antony Widoff, John Offenhartz, and David Zicarelli. Zicarelli recently formed a new software company called Cycling '74, which has released *M* 2.5, intending to "introduce it to a new generation of musicians as well as to provide *M*'s old friends with a new version."

Old fans of *M* will feel right at home with *M* 2.5, which looks and feels much the same as its predecessors. Those of you who missed the program the first time around should prepare for a fascinating experience.

### OVERVIEW

If I were reviewing a sequencer, I wouldn't need to begin by discussing what a sequencer is. *M*, however, is so different from most music software that the question "What *is M*, anyway?" is a legitimate one.

One answer is that *M* is an environment for interactive composition in which you work with your computer to shape basic musical material into a finished composition. In this process, you think more about the probability that some type of event will occur than you do about what happens from note to note. For example, you might specify that an ordered set of pitches be played but also instruct *M* to omit notes at random 25 percent of the time. That's the "composition" part. If you don't like the way the result sounds, you change the percentage while *M* plays—that's the "interactive" part.

Another answer is that *M* is a tool for structured improvisation. Extending our example, you could set up several "presets" in which the pitch material stays the same, but the percentage of omitted notes varies. You could then perform a piece in which you select presets, letting *M* decide exactly which notes to skip or play. No two performances would be exactly the same. In this situation, you function as an improvising conductor, triggering the performance of musical material whose general character is predictable but whose exact minutiae are unknown. If you were ever lucky enough to watch the late Frank Zappa direct a group of improvisers with his infamous hand signals, you will understand what this kind of performance can be like at its best.

Still another way of looking at *M* is as an "idea generator." In this mode, you might set up an improvisation as above and record the improv to a MIDI file. Then, importing the result into a sequencer, you could listen critically and edit, refine, and tweak *M*'s output into a polished piece. The result could be something bearing little resemblance to what *M* delivered initially.

If you're used to composing more traditionally, learning *M* may be challenging because it offers so many new and different ways to think about creating music. Fortunately, *M*'s 191-page manual is a model of clarity. Much of it is devoted to tutorials that take you through every aspect of the program.

### SETTING THE STAGE

Composition with *M* falls into three stages. In the initial stage, data entry, you create collections of notes, chords, and rests, called Patterns. In the next phase, setup, you manipulate various parameters as the Patterns play. You hear the results of your actions immediately as you are performing them. In the third, or structural, phase, you configure larger-scale musical changes, possibly automating them or capturing the output to a file.

*M*'s musical data flows through four Voices. Patterns are the source material of each Voice, and each Voice repeats its Pattern cyclically, much like a looping sequencer track. But a Voice can also transform its data in some fashion, which varies the repetitions.

Each Voice has several Variables. These Variables perform musical transformations, such as transposition or note reordering, on a Pattern while it's playing. Each Variable has six Positions, which store a set of values for the Variable. (You can think of the Positions as presets.) By activating different Positions, you change the settings of all

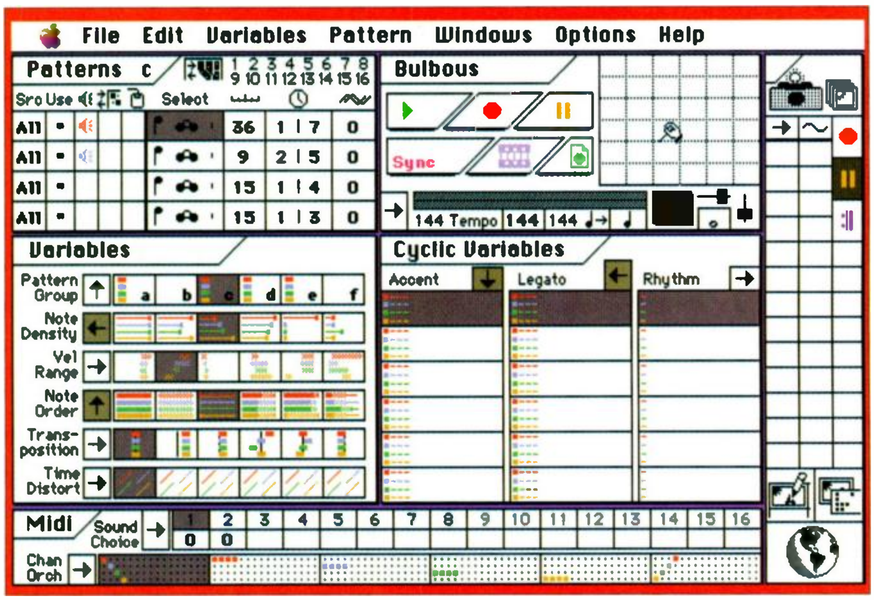



**FIG. 1: *M*'s six main windows are (counterclockwise): Patterns, Variables, MIDI, Cyclic Variables, Snapshots, and Conducting. Note the Conducting grid and Baton cursor in the Conducting window.**

For manufacturer information, please see Contact Sheet, p. 227.

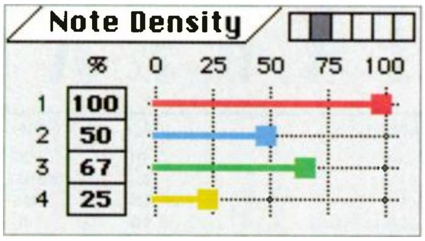



**FIG. 2: The Note Density Edit window determines the probability that a Voice will be heard.**

Voices, instantly changing the musical texture.

The software has higher-level structuring features, too. You can "conduct" *M*'s tempo, the positioning of groups of Variables, and other parameters. You can also record the state of the program in Snapshots and automate a sequence of Snapshots via a Slideshow.

*M*'s live-performance features include the Input Control System, which maps MIDI notes to control functions, and Drum Machine Recording, which allows a Pattern to be continuously rerecorded while being played back and modified by the Variables.

## USER INTERFACE TOUR

*M* presents a compact and elegant real-time control panel consisting of six main windows (see **Fig. 1**). Unlike conventional Mac windows, *M*'s windows can't be closed. This feature protects you from accidentally obliterating part of your "band" during a concert and exemplifies the care that went into the interface design. Another thoughtful touch is the No Zoom Rects option. If enabled, this option blocks the drawing of the traditional "zoom box" when editing windows are opened. This yields a small efficiency improvement—but possibly a critical one in live performance.

Here's an overview of the other main work areas:

The Patterns window has four rows of controls, one row per Voice. Each row has controls for playback, record, and timing functions. Patterns can be selected for editing from the Patterns window.

The Conducting window contains the filename (in this case, "Bulbous") and the controls that affect the program globally. These include transport and tempo controls, as well as the Conducting Grid.

The Variables window has six Variables in a grid with six rows. The rectangles in each row represent the six possible Positions for each Variable. There is always one active Position for each Variable. As soon as you activate a Position, its values become current, and all Voices playing are affected immediately. You can change the values of most Variables in the Edit windows (see **Fig. 2**), which differ according to the Variable. Each control element is efficient and appropriate to its purpose.

The Cyclic Variables window, in the center of the display, contains three more Variables: Accent, Legato, and Rhythm. Cyclic Variables represent repetitive patterns of parameter values.

The MIDI window is used for sending Program Changes and routing Voices to output channels.

The Snapshot window contains automation controls that let you create and execute captured combinations of screen controls, or Snapshots. You can also record and play back sequences of Variable changes and Snapshots, which are called Slideshows.

*M*'s user interface has withstood the test of time remarkably well. It is no easy task to design a user interface that services user control gestures and screen updates while the program makes decisions about what to play and generates several MIDI streams. *M* was originally designed to handle these tasks on a considerably less powerful platform than the contemporary Power Mac—there were no CPU cycles to waste on the gaudy three-dimensional, multicolored screen widgets we're used to nowadays. Consequently, *M*'s user interface has a refreshing austerity.

## PATTERNS

*M*'s Patterns have up to 999 steps, each of which can store a note or chord (a step can also represent a rest). Steps have only a note value—the duration, Velocity, and other event parameters are determined by the Voice's Variable settings at playback time. A Pattern is a source of data from which a stream of notes can be generated, and that stream typically undergoes radical transformations before you actually hear it.

Patterns are organized into six Pattern Groups, each of which is a configuration of four Patterns assigned to four Voices. To record, edit, or play a particular Pattern, you activate one of the six Positions of the Pattern Groups Variable. The Patterns of the active group then become available through the Patterns window.

The Patterns window contains controls for recording patterns from a MIDI device. To record, you record-enable a Voice, select an input channel, and set recording modalities. Record mode determines whether played chords will be broken up into single notes, stored as chords, or built up note by note. Insert mode determines whether new notes will overwrite old ones, will be inserted mid-Pattern, or will be overdubbed. Drum Machine mode is a loop-recording mode.

You can also enter data directly into the Pattern Editor (see **Fig. 3**). To enter notes or chords, you scroll to the desired pitch/step rectangles and click on them. The Pattern Editor supports standard edit functions—region selection, erasure, deletion, insertion, cut, copy, and paste—but not undo. Transposition, augmentation, retrograde, rotation, "scrambling" (randomization), and other musical operations are also available.

You can import data into a Pattern Group from a MIDI file. The import process converts a conventional sequence into a collection of notes, rests, and chords without durations. The resultant Pattern may distort the rhythm of the input sequence. To import and play a sequence as accurately as possible,

**(continued on p. 228)**

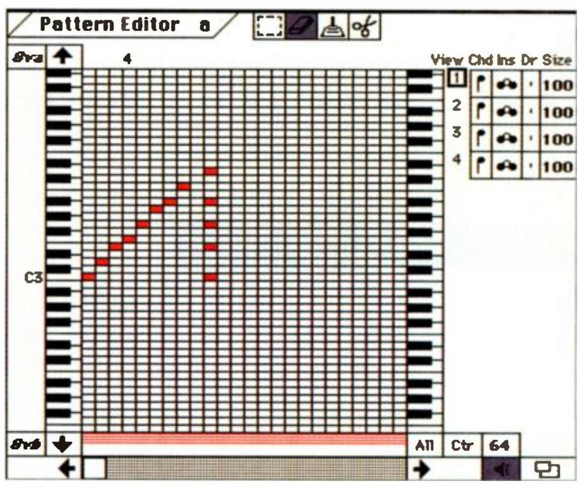



**FIG. 3: *M*'s Pattern Editor window functions like the piano roll in a MIDI sequencer. The Pattern shown here consists of a C Mixolydian scale, followed by a rest, followed by a C9 chord.**

# ADVERTISER INDEX



## RATE THE ARTICLES IN THIS ISSUE! February 1999

We want to know what you think of the articles in *Electronic Musician*! Now you can use your reader service card to give us feedback about EM's editorial coverage. We have assigned a rating number to each of the main articles in this issue. Please select a rating for each article and circle the appropriate number on your reader service card:

**Please select ONE rating number per article**

| | Very Helpful | Somewhat Helpful | Not Helpful | Didn't read |
|---|---|---|---|---|
| A. "Kickin' It," p. 32 | 701 | 702 | 703 | 704 |
| B. Production Values: "To Sir with Love," p. 54 | 705 | 706 | 707 | 708 |
| C. Cover Story "Electronic Euphoria," p. 74 | 709 | 710 | 711 | 712 |
| D. Stage View: "Playing to the Back of the House," p. 100 | 713 | 714 | 715 | 716 |
| E. Performing Musician: "At Home in Your Range," p.144 | 717 | 718 | 719 | 720 |
| F. Operation Help: "Expert Advise," p. 152 | 721 | 722 | 723 | 724 |

## EM Advertiser Sales Regions and Representatives

**Joanne Zola**
Northwest Sales Associate
(510) 653-3307
joanne_zola@intertec.com

**Erika Lopez**
Southwest Sales Manager
(310) 207-8222
erika_lopez@intertec.com

**Julie Clark**
Midwest Sales Associate
(616) 687-8848
julie_clark@intertec.com

**Joe Perry**
East Coast Sales Manager
(770) 343-9978
joe_perry@intertec.com

**Robin Boyce-Trubitt**
Classifieds Sales Manager
(800) 544-5530
robin_boyce@intertec.com


# CONTACT SHEET

**A GUIDE TO THE COMPANIES AND ORGANIZATIONS MENTIONED IN THIS ISSUE OF ELECTRONIC MUSICIAN**

## Kickin' It
**pp. 32–52**

**AKG Acoustics** tel. (615) 360-0499; fax (615) 360-0275; e-mail akgusa@harman.com; Web www.akg-acoustics.com

**Audio-Technica U.S., Inc.** tel. (330) 686-2600; fax (330) 686-0719; e-mail pro@atus.com; Web www.audio-technica.com

**Audix Corporation** tel. (800) 966-8261 or (503) 682-6933; fax (503) 682-7114; e-mail info@audixusa.com; Web www.audixusa.com

**beyerdynamic, Inc.** tel. (516) 293-3200; fax (516) 293-3288; e-mail salesUSA@beyerdynamic.com; Web www.beyerdynamic.com

**Electro-Voice, Inc.** tel. (800) 234-6831 or (616) 695-6831; fax (616) 695-1304; e-mail mtorlone@eviaudio.com; Web www.electrovoice.com

**Sennheiser Electronic Corp.** tel. (860) 434-9190; fax (860) 434-1759; e-mail info@sennheiserusa.com; Web www.sennheiserusa.com

**Shure Brothers, Inc.** tel. (800) 25-SHURE or (847) 866-2200; fax (800) 34-SHURE or (847) 866-2279; e-mail sales@shure.com; Web www.shure.com

## Electronic Euphoria
**pp. 74–86**

**dbx** tel. (801) 568-7660; fax (801) 568-7662; e-mail customer@dbxpro.com; Web www.dbxpro.com

**Drawmer/Transamerica Audio Group (distributor)** tel. (805) 241-4443; fax (805) 241-7839; e-mail transamag@aol.com; Web www.proaudio.co.uk/drawmer.htm

**Joemeek/Peninsula Marketing, Inc. (PMI) (distributor)** tel. (310) 373-9129; fax (310) 373-4714; e-mail themeekman@joemeek.com; Web www.joemeek.com

**Roland Corporation U.S.** tel. (323) 685-5141; fax (323) 721-4875; Web www.rolandus.com

## The Well-Tempered Studio
**pp. 88–98**

**Acoustical Solutions** tel. (800) 782-5742 or (804) 346-8350; fax (804) 346-8808; e-mail info@acousticalsolutions.com; Web www.acousticalsolutions.com

**Acoustic Sciences Corporation (ASC)** tel. (800) 272-8823 or (541) 343-9727; fax (541) 343-9245; e-mail info@tubetrap.com; Web www.tubetrap.com

**AcoustiSoft** tel. (800) 301-1423; fax (905) 721-7540; e-mail sales@etfacoustic.com; Web www.etfacoustic.com

**JBL (Sia Software Company)** tel. (800) 852-5776 or (818) 894-8850; fax (818) 830-1220; e-mail siasoft@aol.com; Web www.jblpro.com

**KB Acoustics** tel. (541) 935-7022; e-mail programs@kbacoustics.com; Web www.kbacoustics.com

**Liberty Instruments** tel./fax (513) 755-0252; e-mail bwaslo@one.net; Web www.libinst.com

**Pilchner Schoustal Architectural Acoustics, Inc.** tel. (416) 868-0809; fax (416) 861-0620; e-mail acoustic@pilchner-schoustal.com; Web www.pilchner-schoustal.com

**RPG Diffusor Systems, Inc.** tel. (301) 249-0044; fax (301) 249-3912; e-mail info@rpginc.com; Web www.rpginc.com

## Desktop Musician: Steal the Scene with SMFs
**pp. 112–118**

**Apple Computer** tel.(800) 538-9696 or (408) 996-1010; fax (408) 974-6615; Web www.apple.com

**Kawai America Corp.** tel. (310) 631-1771; (800) 546-9740; fax (310) 604-6913; e-mail kawaips@aol.com; Web www.kawaius.com

**LiveUpdate** fax (978) 470-3338; e-mail salesdesk@liveupdate.com; Web www.liveupdate.com

**RealNetworks** tel. (206) 674-2374; fax (206) 674-2698; Web www.real.com

**Roland Corporation U.S.** tel. (323) 685-5141; fax (323) 721-4875; Web www.rolandus.com

**Sony Corporation of America** tel. (800) 686-SONY or (201) 930-1000; fax (201) 930-7633; Web www.sony.com

**Tascam/Teac** tel. (323) 726-0303; fax (323) 727-7635; faxback (800) 827-2268; Web www.tascam.com

**Tran Tracks, Inc.** tel. (800) 473-0797 or (973) 383-6691; fax (973) 383-0797; e-mail trantracks@crystal.palace.net; Web www.trantracks.com

**Yamaha Corporation of America** tel. (714) 522-9011; fax (714) 739-2680; e-mail info@yamaha.com; Web www.yamaha.com

## Recording Musician: Recording Resonator Guitars
**pp. 120–128**

**AKG Acoustics U.S.** tel. (615) 360-0499; fax (615) 360-0275; e-mail akgusa@harman.com; Web www.akg-acoustics.com

**Highlander Musical Audio Products** tel. (888) 658-1819 or (805) 547-1410;fax (805) 547-1228; e-mail hmap@highlanderpickups.com; Web www.highlanderpickups.com

**Lawson Microphones** tel. (615) 269-5542; e-mail lawson47@edge.net; Web www.lawsonmicrophones.com

**National Reso-Phonic Guitars** tel. (805) 546-8442; fax (805) 546-8430; e-mail info@nationalguitars.com; Web www.nationalguitars.com

**Neumann USA** tel. (860) 434-5220 or (860) 434-9190; fax (860) 434-3148; e-mail neumlit@neumannusa.com; Web www.neumannusa.com

**Regal Guitars/Saga Musical Instruments (distributor)** tel. (650) 588-5558; fax (650) 871-7590; e-mail saga.music@juno.com

**Royer Labs** tel. (818) 760-8472; fax (818) 760-8864; e-mail info@royerlabs.com; Web www.royerlabs.com

## Square One: Get in Sync
**pp. 130–136**

**Alesis Corporation** tel. (800) 525-3747 or (310) 255-3400; fax (310) 255-3401; e-mail alecorp@alesis1.usa.com; Web www.alesis.com

**Cakewalk Music Software** tel. (888) CAKEWALK or (617) 441-7870; fax (617) 441-7887; e-mail sales@cakewalk.com; Web www.cakewalk.com

**Digidesign** tel. (800) 333-2137 or (650) 842-7900; fax (650) 842-7999; e-mail prodinfo@digidesign.com; Web www.digidesign.com

**Mark of the Unicorn, Inc. (MOTU)** tel. (617) 576-2760; fax (617) 576-3609; e-mail info@motu.com; Web www.motu.com

**Opcode Systems** tel. (650) 429-2400; fax (650) 856-0777; e-mail info@opcode.com; Web www.opcode.com

**Tascam/Teac** tel. (323) 726-0303; fax (323) 727-7635; faxback (800) 827-2268; Web www.tascam.com

## Operation Help
**pp. 152–162**

**Alesis Corporation** tel. (800) 525-3747 or (310) 255-3400; fax (310) 255-3401; e-mail alecorp@alesis1.usa.com; Web www.alesis.com

**Roland Corporation U.S.** tel. (323) 685-5141; fax (323) 721-4875; Web www.rolandus.com

(continued on p. 233)

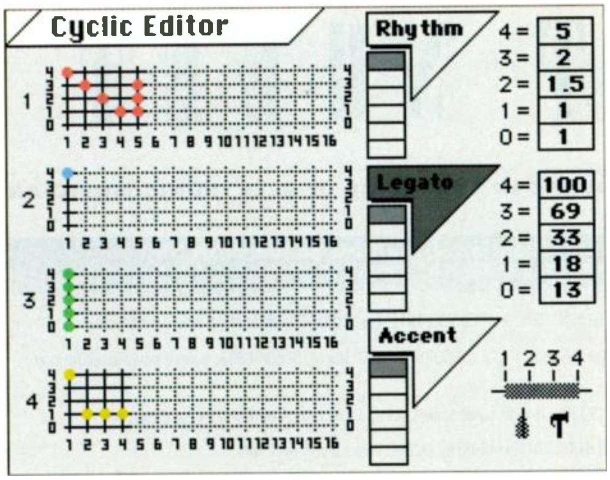



**FIG. 4: The Cyclic Editor window can be used to add expression to a Pattern. The Legato Variable is selected for editing above.**

**(continued from p. 225)**

an Import As Sequence option lets you read in a MIDI file that will be played back as is along with the Voices.

The Pattern record/edit features are pretty straightforward. If you've ever used the step-record or piano-roll editing functions in a sequencer, you'll be quite comfortable with *M*'s equivalents. I'd like to see at least one level of undo implemented in the Pattern Editor, though. Its absence can hamper the work process.

## PLAYBACK, TEMPO & TIMING

The Conducting window contains Start, Stop, and Pause playback controls. There's also a Sync button, which resets all Voices to the beginning of their Patterns. This can be useful if an edit throws Voices out of sync. The Sequence Play Enable button turns on playback of an imported sequence.

The Conducting window's Tempo controls set *M*'s internal clock speed. All Voices play in relation to the master Tempo, which is always expressed in quarter notes. *M* can slave to, or send, MIDI Clock.

Each Voice has a Time Base control, which sets the basic pulse of the Voice relative to the master Tempo. The Time Base consists of two values that express a fraction of a whole note. If a Voice's Time Base is set to 1/4, 2/8, or 4/16, for example, it plays in quarter notes. A value of 1/6 would yield triplet quarter notes, 3/4 would give dotted half notes, and so on. Each Voice also has a Phase control, which can delay a Voice by up to 199 ticks of the master clock.

Time Base can be used to establish a feeling of meter in *M*. You can set up a 12/8 feel, for instance, by setting the Time Base of one Voice to 1/4 and another to 1/12 (eighth-note triplets). You can also set up intricate polyrhythmic relationships between Voices (give 1/11 versus 1/7 a try). Note, however, that *M*'s 96 ppqn timing resolution is low by today's standards, which limits the polyrhythmic possibilities in comparison with many other products. Most sequencers, for example, now feature 480 ppqn resolution or better. Cycling '74 plans to improve the resolution in *M*'s next release.

## VARIABLES

Whether you use *M* to compose, improvise, or run a bank of fog machines, you'll be busy tweaking Variables. Each Variable presents a different Edit window, with appropriate controls. **Figure 2** shows an example, the Note Density window. The values for Voices 1 through 4 are displayed and entered in the boxes on the left, which are called numericals. The numerical, a screen control that users of Opcode software will be familiar with, is one of Zicarelli's best inventions. You simply position the cursor in the box, and press the mouse. If the mouse is positioned in the lower half of the box, the value increments; otherwise it decrements. If the mouse is moved vertically, the numerical acts as a virtual slider. You can also use the horizontal sliders to set the value for any voice. The six Edit Selector boxes near the window's title bar let you edit another Position of the Variable.

Some Variables have a pretty straightforward function. The Orchestration Variable, for example, is a grid that enables you to route Voices to any or all output channels. The Transposition Variable transposes a Voice relative to C3. (A value of D3 would transpose by a whole step up, C2 would transpose an octave down, and so on.) The Pattern Group Variable assigns groups of Patterns (labeled *a* through *f*) to the Voices.

The meaning and effect of the Note Density, Note Order, and Time Distortion Variables may be less obvious. The Note Density Variable is a tool for random variation of rhythm or texture. It controls the probability that a Voice will play. At 100 percent, the Voice will play every note in its pattern. At 25 percent, the odds are 75/25 in favor of "playing" a rest instead of a note.

The Note Order Variable lets you resequence the notes that a Voice plays. There are three types of ordering: Original Order means that the Voice will play back its Pattern unaltered; Cyclic Random means that a randomly scrambled copy of the Pattern will be played repetitively (you can "rescramble" it as desired); Utterly Random means that *M* will select steps randomly from the Pattern. Keep in mind that, because Patterns may contain rests, the Note Order Variable can affect rhythm as well as pitch.

Ordering types can be combined. For example, you could mix 90 percent Original Ordering with 10 percent Utterly Random. This would sound like a slightly disturbed version of the original Pattern, with occasional notes or rests out of sequence. You might also use a Variable to control a percussion setup in which each MIDI note triggers a different timbre. I experimented with this and found it an interesting application. As you can imagine, many permutations are available.

The Time Distortion Variable creates rhythmic effects by varying the rate at which a Voice's clock "ticks." You control the variation by specifying a unit of time, such as a whole note. Then you draw a Time Map, which is a simple linear representation of how the Voice's speed will fluctuate over that unit of time. The voice will then speed up and slow down, "catching up" with itself periodically. Time Distortion can create arrhythmic textures or just imitate a musician whose time is really bad!

I had difficulty creating a musically compelling rubato with Time Distortion. The deviations produced by the Time Map didn't seem subtle enough (possibly due to *M*'s low timing resolution). Also, the cyclic repetition undercut the expressive purpose of rubato. A musician probably wouldn't play rubato for 16 bars, followed by *exactly the same* rubato for the next 16.

Finally, there is the Velocity Range Variable, which works in conjunction with the Accent Variable. I'll discuss this option in the next section.

## CYCLIC VARIABLES

The Cyclic Variables allow you to set up repeating cycles of values that influence the Velocity, articulation, and step duration of each note played. Cycles can be up to 16 steps long, and each step can store a single value or a range from which a value is selected at random.

The Accent Variable selects MIDI Velocity values from a range that is defined by the Velocity Range Variable. The Legato Variable selects values that determine a sustain time for each note, expressed as a percentage of the note's nominal duration. The Rhythm Variable selects values that multiply the step duration (as set by the Time Base) of each Voice.

You access Cyclic Variables through the Cyclic Editor window. **Figure 4** shows the Cyclic Editor with the Legato Variable selected. The window has an Editing Grid for each Voice, and the Grid sets the number of active steps in the Variable's cycle. Colored dots on each active step represent the values of that step.

The Cyclic Variables are essential for livening up pattern-based music. Suppose you have a rather dull bass line of eight notes. If you superimpose a cycle of seven accents on it, you now have a sequence that repeats itself every 56 (8 × 7) events. If you now superimpose a cycle of six articulations, you have a 336-event (8 × 7 × 6) sequence. If you've ever been attracted to so-called "minimalist" music (such as Steve Reich's compositions of the '60s and '70s), you may find yourself hypnotized by the phase patterns you can set up with the Cyclic Variables.

This simple example only begins to suggest the potential of the Cyclic Variables. By randomizing a few of the accent steps and lowering the Voice's Note Density so that the bass line lays out every so often, you could have quite a groove going. Just try asking your bass player to do something like *that*.

*M*'s nine Variables provide an effective set of controls for generating a musically varied and expressive stream of MIDI events. These are almost exclusively note events, of course; if you want other types of events, like Continuous Controllers, you can use the Echo-Thru feature, or you can capture a MIDI file and add the controllers in another program. Would it be desirable to include additional Variables that could generate Continuous Controller messages? Today's powerful CPUs could almost certainly handle the load. But if I were a live performer, I'd want to avoid dealing with the additional complexity. Those who feel otherwise may want to contact Cycling '74.

**M 2.5**
**Minimum System Requirements**
Power Mac or Macintosh 680X0; 2 MB RAM; Mac OS 7.0; OMS 1.2 or higher, QuickTime Musical Instruments (QuickTime 2.5 or higher required) or Apple MIDI Manager

## CONDUCTING AND MOVIES

Conducting is a way to change the state of several Variables or other program parameters simultaneously. You do this by moving the Baton in the Conducting Grid. The Baton can be controlled by the mouse or by selected MIDI controllers.

Each Variable has a Conducting Arrow. The direction of the Conducting Arrow determines the corresponding axis of Baton motion in the Conducting Grid. For example, if a Variable's arrow points up, upward Baton motion will advance the Position; if it points down, downward motion will decrement the Position. You can also conduct Tempo or the execution of Snapshots.

Conducting has two variants: in Continuous Conducting mode, the Baton varies Velocity or articulation smoothly; in Automatic Conducting mode, the Baton is moved to random positions in the Conducting Grid. You set how far the Baton will jump in either direction, as well as the time interval between jumps. Then you are free to walk the dog or cook dinner, secure in the knowledge that *M*'s Robot Conductor is in charge.

The Hold/Do feature is another way to reposition several Variables. When you click on the Hold/Do button (the camera/slides icon at the top of **Fig. 5**), you can select new Positions as desired, but the changes are put "on hold." When you click the Hold/Do button again, all the selected Variables change Position.

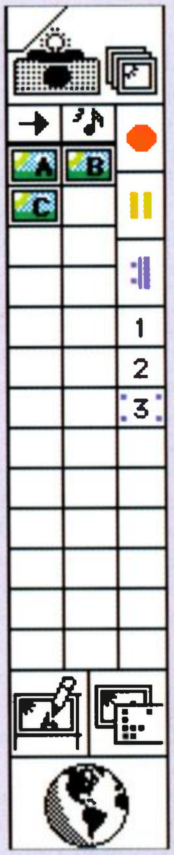



**FIG. 5: *M*'s Snapshot window includes a Hold/Do button (camera/slides icon) in the top area, Slideshow numbers in the rightmost column, and Snapshot icons in the left columns.**

The Snapshot feature stores a record of a group of screen controls. A Snapshot can store the Positions of Variables, Voice controls such as Time Base, and many other parameters. Making a Snapshot is similar to Hold/Do, but the parameter settings are stored in one of 26 locations represented by icons in the Snapshot window. In **Figure 5**, the Snapshot window contains three Snapshots, labeled A through C. To execute a Snapshot, you click on the appropriate letter, and the parameters change to the saved state.

A sequence of Snapshot executions, called a Slideshow, can be recorded and played back using the transportlike controls in the Snapshot window. You can store up to nine Slideshows. In **Figure 5**, the Snapshot window contains three Slideshows labeled 1 through 3.

If you need to export an *M* performance to another program, you can save *M*'s output to a Movie (MIDI File).

## MIDI MATTERS

*M* 2.5 has overcome a major limitation of previous versions by adding compatibility with OMS. In addition, QuickTime Musical Instruments are now supported.

*M* has 16 input/output "channels," which combine a device (or port) and a channel. With OMS installed, you can assign any of *M*'s 16 output channels to an OMS device and a channel within that device. Input channels can be assigned to any OMS controller device. OMS simplifies setup and offers great flexibility—I recommend it.

However, you can bypass MIDI altogether by assigning channels to QuickTime Musical Instruments. *M* also supports the obsolete Apple MIDI Manager and even has an internal MIDI driver for older 680X0 Macs.

Having established channel assignments, you can route each Voice to any combination of channels. You can also send Program Changes via the Output Choice numericals. *M* doesn't support Bank-Select commands, so you're out of luck if your synthesizer has more than 128 programs. I found this limitation rather annoying. Fortunately, Cycling '74 plans to fix this problem in the near future.

*M*'s Input Control System allows a performance to be controlled from a MIDI keyboard. When Input Control is enabled, incoming MIDI notes will drive playback, tempo, Variable Positions, Time Base, and many other functions. The mapping of MIDI notes to functions is fixed by *M;* it is not user-definable.

## M-RELATED LISTENINGS AND READINGS

Several recordings featuring music made with *M* are available from the Electronic Music Foundation (EMF). Joel Chadabe's *After Some Songs* (Deep Listening #DL100) consists entirely of *M* improvisations. *The Composer in the Computer Age-III* (CDCM #CE119) includes Cindy McTee's *M Music. M* also plays a role in Bruno Spoerri's *Movin' On* (Turicaphon #TU100). You can order these discs from the EMF Web site at www.emf.org.

If you want to read more about *M*, the EMF also has Joel Chadabe's excellent *Electric Sound: The Past and Promise of Electronic Music* (Prentice-Hall 1997), which discusses the origins of *M* and other interactive and algorithmic composition software. David Zicarelli's article, "M and Jam Factory" (*Computer Music Journal*, vol. 11, number 4, MIT Press) is also well worth reading, especially if you're interested in good user-interface design. Robert Rowe's *Interactive Music Systems* (MIT Press 1993), a fairly technical discussion of real-time music software, mentions *M* briefly.

And finally, don't neglect Cycling '74's Web site (www.cycling74.com), where you'll find a free demo and an interesting FAQ by Zicarelli.

## CONCLUSIONS

Some years ago I attended a lecture/demo on *M* by Joel Chadabe. During the question-and-answer session that followed, there was a heated discussion as to which types of music *M* was good for. Chadabe, as I recall, said that *M* was a "general-purpose compositional tool." An opponent maintained that *M* was "style specific," with a bias toward experimental and aleatoric styles. "You'll never get a Bach fugue out of *M*," he maintained. "You'll never get a *Bach* fugue out of *anyone*," countered another participant.

Indeed, it is difficult to imagine anyone producing convincing Baroque counterpoint with *M.* In my opinion, the program is very much informed by postwar experimental schools of artistic thought. It embodies a compositional or performance process in which some elements of a piece are not known until they happen—a process like those used by composers as diverse as Stockhausen, Boulez, Cage, and Crumb, but which probably wouldn't appeal to a risk-averse producer of commercial jingles.

Does this mean that every piece constructed in *M* will sound like the above composers? I don't think so. You can certainly use *M* to generate atonal, rhythmically irregular, postserial textures if that is your inclination, but *M* is not limited to that type of music. The program is a natural for repetition-based tonal genres such as New Age or minimalist. Combined with more conventional material in a sequencer, *M*-generated percussion tracks can spice up a funk groove, too. Chadabe's *M* productions are quite jazzy.

One of my favorite applications for *M* is not for composition per se: I like to use it as a sort of timbre tester. Sometimes I develop synthesizer patches with no particular purpose in mind; *M* is useful for exercising such patches in various ways—trying them out in different registers, Velocities, articulations, speeds, densities, and combinations. Using *M* this way helps me imagine what musical context my sounds might function in. Sometimes the germ of a musical idea, or at least an enjoyable surprise, will emerge from this process.

*M* is often described as an "intelligent instrument." Like any instrument, it requires study and practice before virtuoso results can be achieved. Don't be intimidated, though; *M* is fun to learn and fun to work (or play) with. Even composers who like to control every last detail of their work (as I do) have been known to use *M* just to break out of a rut.

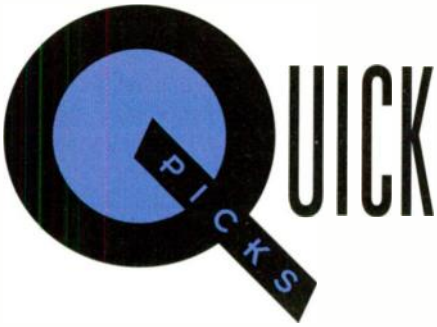
*M* is a unique and powerful instrument for composition and experimentation, as well as something of a software classic. It costs only 74 bucks, which is about half its original price. How can you lose?

**CYCLING '74**
M 2.5 algorithmic composition software
$74

| | |
|---|---|
| FEATURES | 5 |
| EASE OF USE | 4 |
| DOCUMENTATION | 4.5 |
| VALUE | 4.5 |

**PROS:** Provides a unique environment for interactive composition. Superb user interface. Excellent documentation. Affordable price.
**CONS:** Low timing resolution. Limited MIDI Program Changes (no bank select). Reads and writes only Type 0 MIDI files.
**CIRCLE #444 ON READER SERVICE CARD**

**John Duesenberry**'s *electronic music is available through the Electronic Music Foundation. Check the EMF catalog at www.emf.org.*

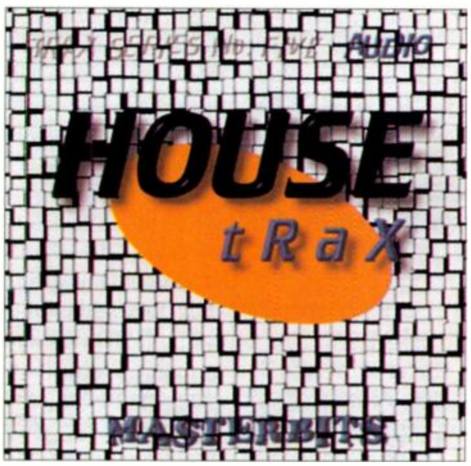



## MASTERBITS

### House Trax

*By Jeff Obee*

Dance aficionados the world over shake their bodies to the pumping house groove, and there are many sample discs that focus on this genre. Masterbits has produced a number of quality dance-oriented CDs, and *House Trax* ($69, audio CD; $39, WAV file CD; $129, Akai CD-ROM, available this month) is its latest addition.




**The basic elements of house-style grooves reside on Masterbits' House Trax sample CD.**

### A Look at the Loops

There's a strong emphasis here on drum and percussion loops, for obvious reasons—just try dancing without them. *House Trax* has 267 house drum loops to draw from, so you're sure to find plenty to suit your taste. This form of Euro-House descends from disco, so you'll hear similarities in the drum patterns: swishing hi-hats, upbeat tempos, and bouncy rhythms.

The loops, each two bars in length, are either 120, 125, or 130 bpm and thus fall into a certain danceable "pocket." Many of the loops are alike rhythmically, but the tones are varied, as the kits are equalized quite extensively. Most are processed, electronic-sounding kits with sundry percussion thrown in—conga being the most common—created on an Akai S3000 with sounds from the Masterbits library as well as some of the producers' own.

### The Rest of the Story

The remainder of the disc consists of one-shot samples. The CD flies through 129 drum samples, providing you with a spectrum of colors for drum kits. Kicks, claps, snares, rides, and toms give you the fundamental drum sounds; shakers, congas, finger snaps, and mixed percussion are available for spicing up those sounds.

The Selected House Samples cover much of what you hear in house music. They start with a smattering of analog and FM-type synth basses. Each is sampled at only one note (a low C), which gives you about two usable octaves—not an abundance, but an acceptable range for this type of music.

Following the basses are a few piano, organ, and string patches, which are all given in tritone steps through a five-octave range. The pianos tend toward the synthy Wurlitzer style that often accompanies house, while the organs have a nice grit and bite to them. Some of the strings fall into that '70s string-machine vibe, some sound like quirky early-'80s analog, and others sound FM based; but all are, again, synthy sounding. Twenty talk-box samples end this section for true house flavor.

The Female House Vocals category offers slices of R&B soul from four vocalists who belt out lyrics like "touch me, oh yeah," with some nice harmonized bits thrown in. Typical Housestuff is overflowing with 399 single samples that race by more quickly than the drum samples. These are wavering pads, washes, and other synth chunks that serve as ornaments for your tracks.

### Perspectives

*House Trax* is a solid offering that works well for the specific musical style at which it's aimed. As is my usual reproach, I was unhappy with the documentation: individual tracks are not indexed, and keys aren't provided for the vocal samples or the Typical Housestuff.

Every sampling musician needs a large pool of colors to dip into, and *House Trax* can add workable, if not terribly unique or exciting, choices to any collection. If house is your thing, these Trax will find a happy home in your sampler.

**Overall EM Rating (1 through 5): 3**

**CIRCLE #445 ON READER SERVICE CARD**

## MUSICNET

### AD24 and DA24

*By Erik Hawkins*

MusicNet's AD24 ($749) and DA24 ($549) are each equipped with an ADAT Lightpipe interface and are the first trickle of what will soon be an unstoppable flood of affordable 24-bit converters. MusicNet converters do wonders for old 16-bit ADATs, but they really shine when used with a Lightpipe-equipped sound card and software that can store 24-bit audio files.

### Getting Acquainted

Both the AD24 and DA24 are 8-channel, half-rackspace devices. Each unit has a sample-rate select button (48 or 44.1 kHz), an LED showing the digital lock status, and two LEDs per channel that show when signal is present and when the peak signal level exceeds -2 dBFS. The AD24's clock source is internally switchable between input and output. The AD24 also includes a


**MusicNet's AD24 and DA24 provide an affordable 24-bit solution for any ADAT Lightpipe–equipped recording system.**

calibration mode that keeps the converter's internal modulators and integrators properly matched for optimum performance; use it after the unit has warmed up or after changing sample rates. The DA24 allows you to choose either 24-bit or 16-bit resolution, and it also includes a de-emphasis filter that eliminates the high-frequency boost that is often used with commercial CDs.

The rear panels of both machines are identical: analog connections are made via balanced XLR jacks; word-clock input (AD24) and output (DA24) use standard BNC connectors; a minijack is provided for ground connection; a 15 VAC jack hooks up to a lump-in-the-line power supply. The word-clock connections are an unexpected treat because of the increasing popularity of digital mixers and the general movement toward all-digital systems. Although the DA24 only generates word clock, the AD24 is able to both generate and sync to word clock.

The DA24's dynamic range is 106 dB, with a frequency response of 10 Hz to 20 kHz and a THD of -97 dB (24-bit at 0 dB). The AD24's dynamic range is 117 dB, with a frequency response of 20 Hz to 22 kHz and a THD of -100 dB at -1 dBFS input level. Crosstalk (which MusicNet calls "interchannel isolation") on both units is <110 dB at 1 kHz.

### Sound Proof

These converters sound as solid as they're built. Recordings made with the AD24 and played back through the DA24, regardless of whether the audio was stored at 16- or 24-bits, sounded noticeably better than when using older, 16-bit ADAT converters. When compared to other, more expensive 24-bit systems, it was nearly impossible to pick which one sounded better. The AD24 and the DA24 are worth considering if you're looking for moderately priced, no-frills converters with a Lightpipe interface.

**Overall EM Rating (1 through 5): 4**

**CIRCLE #446 ON READER SERVICE CARD**

## EARTHWORKS

### LAB102

*By Rob Shrock*

The LAB102 ($1,500) from Earthworks is a 2-channel, solid-state microphone preamplifier boasting high-quality specifications as well as features not usually found in affordable preamps. In addition to the balanced XLR inputs on the rear panel, the LAB102 has three separately driven outputs for each channel, for a total of six separate output drivers. The outputs are divided into two categories: stepped and variable.

### Fine Tuning

The stepped outputs (one balanced XLR jack for each channel) are directly linked to gain knobs that are notched in 6 dB increments. This allows for the precise matching of gain settings between the two channels.

The variable outputs (with a balanced XLR and a ¼-inch TRS jack for each channel) are controlled by a continuously variable, smaller knob that adjusts the signal between 0 dB and -20 dB. The variable controls allow for greater precision when setting levels and are designed to be set after the stepped gain knobs are set. The ¼-inch output jacks allow the LAB102 to be easily assimilated into setups that aren't operating at strictly +4 dBu levels.

Each channel has a polarity switch, 48V phantom power, and a standby switch. The standby switch mutes the output at the preamp, allowing for noiseless connection or movement of microphones. Another nice feature is the lock/unlock switch that safely clamps down the connector of the 18 VAC lump-in-the-line power transformer.

### Safe Specs

Like all of Earthworks' products, the LAB102 boasts remarkable specifications: the frequency response is rated at 2 Hz to 100 kHz (± 0.1 dB) with a THD of 0.02 percent at 8 VRMS from 10 Hz to 20 kHz; rise time is a lightning fast ¼ microsecond. Earthworks claims that the LAB102's noise level decreases as you dial in more gain: -125 dBV at 20 dB gain; -133 dBV at 40 dB gain; -136 dBV at 60 dB gain.

While it is certainly true that the LAB102 is quiet and clean, the unit's transparency can also be disconcerting at times. While recording a grand piano with the LAB102 and a pair of Earthworks Z30X microphones, I found myself wanting to add some EQ to the lower mids because the frequency response of the recording path was so flat. As it turns out, I wasn't hearing the coloration of the mic and preamp combinations that I've grown accustomed to, which favor certain frequencies.

The LAB102's transparency is useful when you're looking for the right microphone for a specific application. While auditioning a variety of microphones through the LAB102, I found that, because of the unit's clarity, the character of each microphone came through loud and clear. If you're looking for a noise-free and sonically accurate preamp, the LAB102 delivers in spades.

**Overall EM Rating (1 through 5): 4**

**CIRCLE #448 ON READER SERVICE CARD**

## AMG

### Black II Black, vol. 4

*By Jeff Obee*

I do so love the funk, and AMG's *Black II Black*, vol. 4 (audio; $99.95), gives it up. It's a double CD set with loops, funky dope guitars, brass licks, vocals, boomin' bass, synth effects, and keys: in short, everything you need to create some serious downbeats.

### Give Up the Funk

Thirty-five drum loops get the first CD off to a good start. Each track contains a loop with full instrumentation, followed by versions with one or more elements taken out of the mix. The drum tracks are clearly delineated in the documentation, and bpm settings are given for all. Each track has a distinct and very funky groove, and there is a pronounced tonal variety in the kicks and snares. Turntable scratches are in the mix on some tracks, vinyl crackles on others.

There are 24 tracks of guitar and 15 of bass. (Each track has ten samples.) Again, I heard a variety of guitar sounds, with lots of good tones and effects: the ubiquitous muted Strat sounds, lots of good wah-wah bits, some Wes Montgomery–style octave work, and loads more. The key centers aren't listed, though, so you'll have to figure those out yourself.

The bass samples, on the second CD, are referred to as "Bin Bustin'," although they aren't the deep, subsonic synth bass sounds you'd expect. They are performed by a live bassist who lays down some fine slap and finger funk, and there is some


**Earthworks' LAB102 mic preamp has an extremely flat response and virtually no tonal coloration.**

For manufacturer information, please see Contact Sheet, p. 227.


AMG's *Black II Black*, vol. 4, features the sounds of funk and a wealth of distinct grooves.

slight amp distortion on some tracks (which I assume was intentional). The bass is realistic, but I found anomalies in some of the tones that didn't agree with me. Hey, I'm a bassist, and I have particular tastes in that area.

The brass section is extensive, with hits, swells, slides, licks, and stabs galore, and you also get synth brass licks that are pleasantly plump. The synth samples include some textural synth pads, lead lines, piano, and organ. Many of the comping patterns are straight out of Stevie Wonder. The Worm section of the disc provides a variety of those high, squirrely lines that often appear in hip-hop. I found these samples satisfying, but again, there are no keys given.

Finally, there's a monstrous section of vocal samples. Some of the Big Mama Vocals, which consisted of samples in the "oooh yea" and "gotta keep it movin'" vein, aren't my cup of tea; the vocoded vocals were more to my liking. If you wish to use any of these vocal samples commercially, though, licensing is necessary.

## A Mixed Bag

The lack of specific keys and indexing in the documentation is a disappointment, as is the licensing requirement for the vocal samples. Licensing is completely dependent on the particular project, so you have to contact AMG and work it out with them. However, this is a fun disc; I plugged in my bass and jammed along for hours. The sound quality is top-notch, the performances are low-down funky, and you get quite a variety of fat, contemporary samples that cover all the necessary bases. If funk is your bag, go check it out.

**Overall EM Rating (1 through 5): 3**

**CIRCLE #447 ON READER SERVICE CARD**

## CONTACT SHEET (continued from p. 232)

**Reviews**
**pp. 164–233**

**AMG/Big Fish Audio (distributor)** tel. (800) 717-FISH or (818) 768-6115; fax (818) 768-4117; e-mail info@bigfish.com; Web www.bigfishaudio.com or www.amguk.co.uk

**AnTares Systems/Cameo International (distributor)** tel. (888) 33-CAMEO or (408) 399-0008; fax (408) 399-0036; e-mail sales@cameoworld.com; Web www.antarestech.com

**BitHeadz** tel. (831) 465-9898; fax (831) 465-9899; e-mail info@bitheadz.com; Web www.bitheadz.com

**Cakewalk Music Software** tel. (888) CAKEWALK or (617) 441-7870; fax (617) 441-7887; e-mail sales@cakewalk.com; Web www.cakewalk.com

**Canam Computers/Tracer Technologies Inc. (distributor)** tel. (888) 8TRACER or (717) 843-5833; fax (717) 843-2264; e-mail info@tracertek.com; Web www.tracertek.com

**C-Mexx Software** e-mail cybermexx@t-online.de; Web www.c-mexx.com

**Cycling '74** tel. (415) 621-5743; fax (415) 621-6563; e-mail info@cycling74.com; Web www.cycling74.com

**DigiTech** tel. (801) 566-8800; fax (801) 566-7005; e-mail customer@digitech.com; Web www.digitech.com

**Earthworks** tel. (603) 654-6427; fax (603) 654-6107; e-mail earthwks@jlc.net; Web www.earthwks.com

**Emagic USA** tel. (530) 477-1051; fax (530) 477-1052; e-mail emagic@emagicusa.com; Web www.emagic.de

**Masterbits USA** tel. (888) 678-2487 or (612) 975-9428; fax (612) 975-9429; e-mail mbitsusa@aol.com; Web www.masterbits.de

**Minnetonka Audio Software, Inc.** tel. (612) 449-0187; e-mail info@minnetonkaaudio.com; Web www.minnetonkaaudio.com

**MusicNet/Wave Distribution (distributor)** tel. (973) 728-2425; fax (973) 728-2931; e-mail info@wavedistribution.com; Web www.wavedistribution.com

**Roland Corporation U.S.** tel. (323) 685-5141; fax (323) 721-4875; Web www.rolandus.com

**Steinberg North America** tel. (818) 993-4161; fax (818) 701-7452; faxback (800) 888-7510; e-mail info@steinberg-na.com; Web www.us.steinberg.net

**Yamaha Corporation of America** tel. (714) 522-9011; fax (714) 522-9522; e-mail info@yamaha.com; Web www.yamaha.com

B&H
PHOTO - VIDEO - PRO AUDIO

THE PROFESSIONAL'S SOURCE FOR PHOTO,

FOR ORDERS CALL:
800-947-5509
212-444-6679

OR FAX (24 HOURS):
800-947-9003
212-444-5001

Store & Mail Order Hours:
Sunday 10-5
Monday thru Thursday 9-7
Friday 9-1 • Saturday Closed

On the Web: http://www.bhphotovideo.com

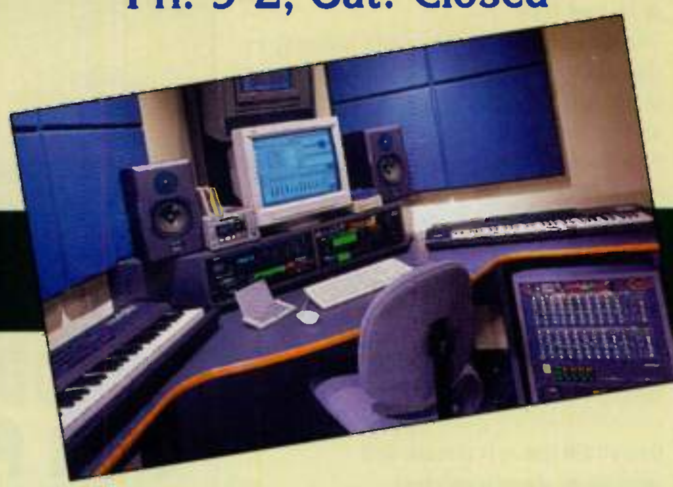
# OUR NEW EXPANDED LOCATION

## 420 Ninth Avenue

*Between 33rd and 34th Streets*

New York, N.Y. 10001

Sun. 10-5, Mon. thru Thurs. 9-7
Fri. 9-2, Sat. Closed



FOR A FREE MONTHLY CATALOG, CALL 800-947-6933, 212-444-6633 OR WRITE

# CLASSIFIEDS

**ELECTRONIC MUSICIAN CLASSIFIED ADS** are the easiest and most economical means of reaching a buyer for your product or service. The classified pages of **EM** supply our readers with a valuable shopping marketplace. We suggest you buy wisely; mail-order consumers have rights, and sellers must comply with the Federal Trade Commission as well as various state laws. **EM** shall not be liable for the contents of advertisements. For complete information on prices and deadlines, call **(800) 544-5530.**

## ACOUSTIC PRODUCTS

**Cloaking Device**
acoustic conditioning systems
- Quick - Easy - Affordable -
modular systems start at 144.00
Start with a Good Sound
770-427-8761
fspace@mindspring.com
www.mindspring.com/~fspace
**Folded Space Technologies**

**WhisperRoom INC.**
SOUND ISOLATION ENCLOSURES
Vocal Booths
Practice Rooms
Broadcast Booths
etc...
PH: 423-585-5827
FAX: 423-585-5831
E-MAIL: whisper@lcs.net
WEB SITE:
www.whisperroom.com
116 S. Sugar Hollow Road
Morristown, Tennessee 37813

**AcousticsFirst™**
Toll Free: 888-765-2900
Full product line for sound control and noise elimination.
Web: http://www.acousticsfirst.com

**SILENT**
58 Nonotuck St., Northampton, MA 01062
INFO (413) 584-7944 • FAX (413) 584-2377
ORDER (800) 583-7174
info@silentsource.com • www.silentsource.com
Acousticore Fabric Panels • Sound Barrier
Isolation Hangers • A.S.C. Tube Traps
Silence Wallcovering • WhisperWedge
Melaflex • S.D.G. Systems • Tecnifoam
R.P.G. Diffusors • Sonex • Sound Quilt

**★ACOUSTIC FOAM★**
Diffusors • Bass Traps • Broadband Absorbers • Sound Barrier & More

**Introducing the Eclipse™ Stand-Mounted Modular Acoustic Environment!**

**Famous artists, motion picture companies, studios & networks choose Auralex time & time again because we make the world's best sound control products regardless of price. We offer free, no-pressure advice & can solve any sound problem. Call us today!**

USERS: James Taylor, Journey, Sammy Hagar, D'arcy of Smashing Pumpkins, Clint Black, Celine Dion, Doane Perry (Jethro Tull), Dennis DeYoung/James Young/Todd Sucherman (Styx), Spyros Poulos (Madonna/Al Dimeola/Kathy Troccoli/Taylor Dayne), George Terry (Clapton), Andrew Gold, Christopher Cross, Mike Wanchic (Mellencamp), Ross Vannelli, Skid Row, Mark Lindsay (Paul Revere & The Raiders), Norbert Stovall (D. Parton/Gatlin Bros./K. Rogers/E. Rabbitt), Bill Kreutzmann (Grateful Dead), Mark Farner (Grand Funk), Bellamy Bros., Wayman Tisdale, Nona Hendryx, John Baxter (Re-Flex), Steve Oppenheimer/Editor/Electronic Musician, Brian Knave/Asst. Editor/Electronic Musician, Michael Molenda/Editor/Guitar Player, A "Phamous" Band, Disney, Michael Cooper/Audio Writer, MUMFORD (A Lawrence Kasdan/Touchstone Film), Bose, LucasArts, Silicon Graphics, Todd-AO, JBL, Shure, Pelonis Acoustics (Skywalker Sound), Bob Hodas, Steven Klein/Fmr. Sr. Eng./Criteria Studios (Clapton/Bee Gees/A. W. B.) & Studio Designer (Don Was/Kenny G/Yab Yum Entertainment/Larrabee), Sear Sound (N. Merchant/S. Colvin), Castle Oaks Prod. (Titanic/Tremors/Wings/21 Jump Street/S. Mendes/J. Feliciano), Sound Kitchen (Frampton/Petra), Digital Insight, AT&T, NASA, Sony, CBS Sports, NBC Sports, ABC TV/Radio, ESPN, MSNBC, Discovery Channel, Bill Fulton (I Am Weasel/Cartoon Network), Premier Percussion, Tom Lavin (Animaniacs, Pinky & the Brain), Berklee College of Music, ShowCo, Opryland, Music Bakery, Audix, Ensoniq, WhisperRoom, Hitachi, Toshiba, 20th Century Fox, Universal Studios, NPR, Warner Bros., Polygram, Fruit of the Loom Country Comfort Tour, Manley Audio Labs, Lucent Technologies, 3BTV, Electroplex, Azden, Macromedia, Adaptec, Martin-Logan, Crawford Post, Pioneer, Carvin, MTX, the voices of McDonald's, Miller Brewing, Sports on Fox and more!

**Auralex** acoustics
www.auralex.com ★ auralex@auralex.com
(317) 842-2600 ★ Fax (317) 842-2760
(800) 95-WEDGE

**ETF** energy time frequency

**ETF** is a software application that runs on your computer using Windows 95/NT, any 16 bit sound card, and an omni-directional microphone.

- **ETF Room Acoustic Analysis**
  measuring - Early and Late Reflections
  - Room Resonance
  - Frequency Response
  - Distortion
  - Reverberation Time Analysis
- **Device Designer** - Design of Helmholtz Resonators and Acoustic Diffusers
- **A primer and tutorial on Room Acoustics**
- **Using *Time Delay Spectrometry* techniques**
- **Used by Professional Studios Worldwide**

The **ETF** System lists for $199.95US Call: 1-800-301-1423
Visit www.etfacoustic.com e-mail: sales@etfacoustic.com

## COMPUTER SYSTEMS

**DUPLICATORS:**

Panasonic 4x8 CD-R bundle..... $295
TEAC 4X12 CD-R bundle.. $349
Yamaha 4x4x16 CD-R bundle.... $449
Plextor 8x20 CD-R bundle... $649
Sanyo 8x20 CD-R bundle... $649
Jaz 2GB + Yamaha 4x4x16 exter. bundle... $899
AFFEX CD Color Printer 1440 dpi.... $1,399
LabelMaker for CD's....... $39

CD-R BLANKS as low as 99¢
also available Ricoh & More

| Replica | | MediaFORM | |
|---|---|---|---|
| SA1000 1CD-R | $989 | CD5900... | $5,499 |
| SA3000 3CD-R | $2,399 | CD3704... | $7,499 |
| SA8000 8CD-R | $4,499 | CD 37068 w/printer... | $15,999 |

FOR CD DUPLICATION... CALL

**Shimad**
1-888-4-SHIMAD
toll free 888-474-4623 PH. 650-493-1234
www.shimad.com FAX: 650-493-1333

## EMPLOYMENT OFFERED

**CAREER CHANGE**
$50k-plus potential. If you enjoy an energetic, goal-oriented approach to business, come join one of America's oldest and most respected music stores. **Caruso Music** is seeking individuals for sales and store management positions. Health benefits, paid vacations, and a professional work environment. Call Richard Caruso, **Caruso Music, Inc.**, 94 State St., New London, CT 06320. (800) 264-6614.
All replies treated confidentially.

**Top Bay Area 24-Trk studio needs young, aggressive staff Engineer for heavy client workload. High stress—low pay. Resumes only to: 444 Natoma St., SF, CA 94103.**

**WANTED:**
**Studio Engineer/Studio Mgr.**
To work in rural Northern Ca.
Studio next to Oregon border near Mt. Shasta, off I-5.
Contact Chuck
(530) 468-5675.

**Do you want a great career?**
Positions Available Now!

**Sweetwater** SOUND INC.
Music Sales, Desktop/Web Design, Sales & Administrative Assistant, Tech Support & More! Contact Kristine!
(800) 222-4700 • (219) 432-8176 • FAX (219) 432-1758 • email: careers@sweetwater.com

**COLORADO LIVING**

The National Digital Television Center in Denver, CO is looking for an audio post engineer with professional experience in audio editing/sweetening for television and film to head up our new world-class digital audio post facility. This is not a job for beginners. The successful candidate must be proficient in Pro Tools-based production and have a working knowledge of digital automated consoles (02R).

Candidate should also be experienced in VO, ADR, MIDI, and ISDN sessions and, of course, have a massive understanding of sync.

Call Laurie Gordon at (303) 466-3655,
weekdays between 8:30 - 10:30 AM MST.
Selected applicant must pass a drug screening and criminal background check.
The National Digital Television Center is an Equal Opportunity Employer.

# CLASSIFIEDS



## EQUIPMENT FOR SALE

# CLASSIFIEDS

## EQUIPMENT FOR SALE

**Sound Deals, Inc.**
Specialists in ***Samplers, Synths, Pro Audio, Analog & Digital Recording, Effects, Drum Machines, Computer Software/Hardware, & more!***
**(800) 822-6434/(205) 823-4888**
*Sound Deals, Inc.*
*230 Old Towne Rd.*
*Birmingham, AL 35216*

**THE CASE SPECIALISTS**
FREE CATALOGUE

**(800) 346-4638**
(516) 563-8326, NY
(516) 563-1390, Fax
Custom or stock sizes.
Our prices can't be beat!
Discount Distributors

**MIDI-KIT CATALOG:** 30 kits, including Programmable Controllers, Relay Driver, Custom Instrument, CV-MIDI, Programmable Transmitter, Data Monitor, MIDI project book, and newsletter for Artistic Technologists. PAVO, Inc. **(800) 546-5461**; www.pavo.com

STICKY mess left on your Mixer/ Patch bays/Cables by adhesive tape? Try *"NO MORE TAPE?"*™ You owe it to yourself & your gear! **Recording Console Concepts** (612) 588-6430
**E-mail:** nomoretape@uswest.net
**Website:** www.nomoretape.com

**Sound thinking**
**Digital Audio: It's What We Do!**
digidesign ALESIS
**...and more**
**541-386-2682**
new! website
www.soundthinking.com

**we've got your n'hummer.™**
**it's simple. it's inexpensive. it removes hum from audio lines without transformers, noise gates or comb filters. guaranteed. numb the hum today! $249.95 for a 2-channel unit.**
roar!
**more info: www.stro-lion-technologies.com 800.567.0881**

**PARTS EXPRESS**
ELECTRONICS & MORE
**FREE CATALOG**
**ASK ABOUT OUR WHOLESALE PRICING PROGRAM!**
- MICROPHONES
- MIC STANDS
- CABLES
- HEADPHONES
- RAW SPEAKERS
- POWERED MIXERS
- PA SYSTEMS
- CABINET & RACK ACCESSORIES
- STAGE LIGHTING
- MIXERS
- POWERED AMPS
- SHOCKMOUNT RACK CASES
- HORN DRIVERS

*Visit our Web Site: www.parts-express.com*
1-800-338-0531

CALL TOLL FREE **1-800-338-0531**
725 Pleasant Valley Dr., Springboro, OH 45066-1158
Phone: 513/743-3000 ◆ FAX: 513/743-1677
E-mail: sales@parts-express.com
*Source Code: EMM*

**How BIG is Yours?**
HARD DRIVES
CD RECORDERS
MEMORY CHIPS
SOUND CARDS
MIDI SOFTWARE
Digital Audio Specialists!
**BIG DISC**
(954) 749-0555
**http://www.bigdisc.com**

**Every** major brand of everything. Millions of dollars of musical gear in stock. **ALTO MUSIC**. Guitars, recording, keyboards, amplifiers, drums, pro sound, new & used. One of the largest selections in the country. We ship everywhere! **(914) 692-6922** • 680 Rt. 211 East Middletown, NY 10940
Ask for Uncle Freddy—He loves ya!


**CL6/1800 Midi Lighting Dimmer**
- 6 Independent Channels of Dimming
- Responds to Midi Notes and Controllers
- Only $449.00 US List Price !

*Ask about our other Midi products of control Valves, Relays, Servo's, Lighting, Analog Keyboards & more!!*
MEDIA Mation
Show Control & Multi Media Systems
2461 West 205th St. B100
Torrance, CA 90501
Phone: (310) 320-0696
Fax: (310) 320-0699
Visa, MC, Discover & AmEx. "Dealer Inquires Welcome"

**OMNIRAX**
STUDIO FURNITURE
**MixStation /O2R**

**Custom Design and Modifications Available**
**Available for All Mackie 8-Bus mixers, & Eurodesk**
**800.332.3393**
FAX 415.332.2607
Outside U.S. 415.332.3392
WWW.OMNIRAX.COM
P.O.Box 1792 Sausalito, CA 94966

**SINGERS!** REMOVE VOCALS
**Unlimited Backgrounds™**
From Standard Tapes, Records, & CDs with the **Thompson Vocal Eliminator™**
Visit our Internet Site at http://www.ltsound.com

**LT Sound**, Dept EM/1
7988 LT Parkway
Lithonia, GA 30058
***24 Hour Demo/Info Line (770)482-2485 - Ext 16***
**Best Vocal Eliminator™, Key Changer, Voice Enhancer!**

**All Major Brands • Lowest Prices • Best Service/Support • Expert Advice**
**DIGITAL PRO AUDIO**
**http://www.digitalproaudio.com/**
**800-240-4079 sales@digitalproaudio.com**
Auto-Tune • V8 • ACID • GigaSampler • Pulsar • Aark 20/20 • Layla • STUDI/O
AKG • dbx • Rode • Lexicon • Cakewalk • ART • Alesis • Tascam • Roland
Mics • Monitors • DAW's • Plug-Ins • DAT's • MDM's • MIDI • EFX • CDR'S

**Make your ad more visible with Color**
**(800) 544-5530**

## EQUIPMENT WANTED

**USED ADATS WANTED**
Easy exchange toward a newer or different format. Everything available. Save **THOUSANDS** when you deal with our 70-year-old company. **UPGRADE TODAY.** Call, fax, or e-mail for details. Worldwide delivery. **CARUSO MUSIC**, New London, CT; sales@caruso.net
**(800) 264-6614 TOLL FREE**
(860) 442-9600
(860) 442-0463 (FAX)
http://www.caruso.net

**(800) 264-6614**
We want your used keyboard, recording, and audio equipment. We'll **airmail** you **$$$** or take your stuff in trade toward over **350 brands** of new and used products. **Come in or do it all through the mail.** 69 years in business. **Worldwide delivery**. Visit our new **22,500 sq. ft.** location. Call, fax, or e-mail us today. **carusomusi@aol.com OR sales@caruso.net.** Visit us at www.caruso.net. Start saving money today. Call **Caruso Music**, 94 State St., New London, CT 06320 USA. Outside of the U.S. call **(860) 442-9600.**

## INSTRUCTION

**BE A RECORDING ENGINEER**

**★ TRAIN AT HOME**
**Easy Home-Study practical training in Multi-track Recording. Join our successful working graduates or build your own studio. Career guidance . . . Diploma . . . Licensed.**
*FREE INFORMATION:*
**Audio Institute of America**
814 46th Ave, Suite F, San Francisco, CA 94121
Or visit us at www.audioinstitute.com

**FREE CATALOG! (800) 275-0797**
Computers • Music • MIDI
Digital Audio • Guitar Catalogs!
Lessons • Books • Tapes • Videos
**CALL PEBBER BROWN**
**(800) 275-0797**
www.angelfire.com/ca2/pebber

**LEARN the ART of RECORDING**
THE RECORDING WORKSHOP
- Short-Term / Affordable Tuition
- Small Classes / 8 studio facility
- On-Campus Housing
- Job Placement / Financial Assist.

*Call Today for a Free Brochure*
**800-848-9900**
**The Recording Workshop**
455-L Massieville Rd Chillicothe OH 45601
ph:740-663-2544 fax: 740-663-2427
email: info@recordingworkshop.com

# CLASSIFIEDS

**Recording Engineer Broadcasting**
Multimedia/Digital/Video/Film
Radio/TV/Sports/News/DJ/Talk Show
No experience required!
**On-the-job-training**
in local major Recording Studios & Radio/TV Stations
Part-time, nights, weekends - Call recorded info line for FREE video
**1-800/295-4433** www.radioconnection.com

Comprehensive Audio Engineering Program
AUDIO RECORDING TECHNOLOGY INSTITUTE
Music Prod. & Digital Recording
Extensive hands-on instruction.
Call (888) 543-ARTI

"I Prefer EM Classifieds because, like our business, they just perform better..."
Anthony Felos
Prophet Disc

## INTERNET SERVICES

**A La Carte Web Design** tm
a division of ROCKA RECORDS
Web Design & Web Hosting for Record Labels, Studios, Musicians, Groups, etc.
Professional HTML, graphic art, animation and recording services for the serious web site owner... to learn more, visit rocka.com.
**Music on the Web** tm
http://musicontheweb.com
MasterCard VISA
Domain & Web Site Hosting
RealAudio·RealVideo· streaming media storage. MP3 file encoding!
Call 1-800-96-ROCKA
www.rocka.com
e-mail: rocka@rocka.com fax: 425-640-6262

## MAINTENANCE SERVICES

**DATs FIXED FAST!**
Expert solutions for all DAT recorder problems
• All makes and models
• Experienced, factory-trained techs
• Fast turnaround
• Repairs, overhauls, parts
*"The DAT doctor is in!"*
NXT GENERATION inc
973-579-4849 FAX 973-579-7571

## PARTS & ACCESSORIES

**Studio Dust Covers**
Covers For Your:
Synths • Mixing Boards
Amps • Sound Systems
Custom Sizes Available
Call Larry LeCover at 1-800.228-3878
or Visit www.lecover.com
The *Le Cover* Co.
1223 Kingston • Schaumburg, IL 60193

Subscription Questions
(800) 843-4086

## RECORDING SERVICES & REPAIRS

GROOVE HOUSE RECORDS
**1000 CD'S** only **$1180**
Includes 3 color CD face, jewel box & shrink wrap. Major label quality. Fast!
1-888-GROOVE-8

**Conversion Media**
Download Our Full Color Catalog
www.conversionmedia.com
CD Cassette Vinyl DVD
Printing Interactive Web
design@conversionmedia.com
**888.999.1760**
Short Run Specialists

MANUFACTURERS OF
**CDs and CASSETTES**
CD REPLICATION -
CASSETTE DUPLICATION -
DIGITAL MASTERING -
GRAPHIC DESIGN -
PRINTING - PACKAGES -
ONE OFF CDS
Ask about our FREE WEB PAGE offer
HTTP://WWW.PB.NET/~CCMUSIC
**800-289-9155**
OUTSIDE U.S. 516-244-0800
C&C MUSIC

**Best Price... Best Service... Period.**
**CD REPLICATION**
• CDs in Retail-Ready Packages
• CDs in Bulk (minimum order only 100 CDs!)
• Vinyl Records, 7 & 12" - colors available!
• Cassette Duplication
Work directly with our factory and save! Call for Free Catalog or Quote: **(800) 455-8555**
http://www.europadisk.com
Major Credit Cards Accepted
EUROPADISK, LTD.

# CLASSIFIEDS

## RECORDING SERVICES & REPAIRS

**Lonely Records**
1 CD only $10
10 CDs only $39
25 CDs only $80
100 CDs only $245
mastering as low as $30
GRAPHIC DESIGN for inserts
1 800 409-8513

**ESP**
*We Anticipate Your Every Need*
**CD REPLICATION**
Cassette Duplication
Graphic Design & Printing
Digital Editing & Mastering
**1-800-527-9225**
(716) 691-7631 • Fax (716) 691-7732

**Best Pick**
**DIGITAL FORCE**
*Feel the Power of Excellence*
(212) 252-9300
www.digitalforce.com
TOTAL CD, CD-ROM, E-CD, & CASSETTE PRODUCTION
149 MADISON AVENUE NY, NY 10016

**It's Your BABY...** *and we deliver.*
CD PACKAGES
500 $750.00
1000 $1,100.00
5000 $4,500.00
Services:
• Mastering
• Design
• Film & Printing
• Web Development
• Enhanced CDs
THE MEDICINE FOR DIGITAL MEDIA
**DIGIDOC** PRODUCTIONS
800-DIGIDOC

**COMPACT DISC**
ALL SERVICES AVAILABLE
• *Highest Quality* • *Low Prices*
• *Superior Service* • *Est. 1986*
• *Member Better Business Bureau*
1-800-900-7995
1-802-453-3334
fax 1-802-453-3343
CD AUDIO • CD ROM
AUDIO & VIDEO CASSETTE
PRINTING / PACKAGING
MASTERING • GRAPHIC DESIGN
**PROTOSOUND** INC.
MASTERCARD • VISA • AMERICAN EXPRESS

**A great deal!**
**Real-time cassettes**—Nakamichi decks, chrome tapes—the best! Album length $1.50/100. On-cass. printing/inserts avail. Grenadier, 10 Parkwood Ave., Rochester, NY 14620. (716) 442-6209 eves.

**ARE YOU GETTING YOUR MONEY'S WORTH?**
CD • CASSETTE • VINYL • DVD
Full-service mastering and post-production
Revolutionary graphic services
Complete internet development and hosting
Digital bin cassette replication
Direct to metal vinyl mastering (HQ180™ Vinyl)
Client graphic proofing on the internet
Digital color prepress and service bureau
Complete custom retail-ready packages
Distribution services are available
500 CD Package $1295.00
1000 CD Package $1895.00
Combo Package $2295.00
Free Web Site With Audio Files
WWW.GATEMUSIC.COM
The Gate Music Services, Inc.
Packages include: packaging, mastering, glass mastering, full color insert, 3 color printing on CD or vinyl, bar code, custom graphic services, film, consultation and much more. No hidden costs, tax, "miscellaneous fees" or "surprise" expenses!
1-800-655-1625 • 510-558-9045 • F 510-558-9504

**AUDIOGRAPHIC MASTERWORKS**
Fast, Full Service CD Packages
Digital Mastering
Custom Graphic Design
Sonic Solutions No Noise™ Audio Restoration
901-821-9099
masterworks@earthlink.net
*Every project is a masterpiece*

## RECORDS, TAPES & CDS

10 CDs • $65
20 CDs • $120
50 CDs • $250
1000 CDs • $1280
Random Access Media
1-800-684-8071

**AMERICA'S BEST CD PACKAGE** HANDS DOWN
*1000 CDs Only $1299 Includes:*
• full color inserts • design • film • reference CD • glass master
• 2 color on disk • shrink wrapped • priority proofs
• manufactured at a Major Manufacturing plant for the *Very Best Quality* • Ready in 3-4 weeks!! No Overruns!!
• ask about our short run CDs • 15 years in business
1000 Bulk CDs With Printing $699
**ELECTRIC** *Records*
**100 Cassettes Only $199**
*includes real-time duplication on chrome tape, full color inserts, design, shrink wrapped & free shipping!*
800-367-1084
www.damall.com/holland/holt.html

Castle Technology, Inc.
Cassette Duplication
C-10...$.74 C-20...$.84 C-30...$.94
Printing—Packaging—Labeling
**Single CDs Starting at $6.95**
(800) 636-4432 or Fax (615) 399-8855
http://members.aol.com/Castletch/castle

**In the Studio?**
**FREE Guide**
Saves You Time and Money!
Contact Us Today:
1-800-468-9353
www.discmakers.com
info@discmakers.com
**DISC MAKERS**
EXPECT MORE

**RETAIL-READY CDs: full color inserts**
25 CD Package....$219
50 CD Package....$349
100 CD Package....$529
*Includes: full color insert, tray card, CD label, jewel case, shrink wrap*
Larger CD quantities available
SIENNA DIGITAL • 1-888-674-3662

# CLASSIFIEDS

# CLASSIFIEDS

## RECORDS, TAPES & CDS

**Universal Media**
CD/DVD Manufacturing Services
25 CDs (Media, Print & Case) $175
1,000 Bulk CDs for $670
Ready in 7 Business Days
Tel. (408) 946-0948 or (408) 973-3514
http://www.syntac.com/um

**DON'T WAIT!**
PERFECT PROMO PACKAGES NOW!!!
50 CD SINGLES $179
150 CD SINGLES $419
INCLUDES GRAPHICS ON DISC AND VINYL SLEEVE
FAST TURNOVER!
CALL
B.R.R. 1-888-554-9334

**1000 CDs: $1150**
4 panel 4/1 insert, 4/0 tray card, 2 color CD, bar code, jewel case, assembly and wrap. From your film and CD-R. Add $245 for film.
1000 Bulk CDs: $690

**500 CDs: $935**
4 panel 4/1 insert, 4/0 tray card, 2 color CD, bar code, jewel case, assembly and wrap. Includes film! From your CD-R and art on disk.
500 Bulk CDs: $500

**CD-Rs:**
**100: $300; 50: $200; 30: $150**
With b/w label, insert, tray card and jewel case. Full color packaging also available. Call for prices.
single CD-R from DAT: $15

**100 Cassettes: $184**
C-45 with b/w j-card, labels and norelco cases. Real-time duplication on BASF Chrome Plus tapes. Full color packaging also available. Call for prices.
50 tapes as low as: $39

Call us at (888) MONKEY-5
THE MONKEYHOUSE
www.themonkeyhouse.com

**DRT Mastering**
the Analog Specialist
Delivering Punch, Warmth & Air - the fat NY/LA sound of custom-built high-performance mastering gear - for your album.
800-884-2576
Free brochure www.drtmastering.com

www.yourmusiccd.com
10 cds - $70
50 cds - $200
Atlanta 678-442-0933
Toll Free 877-442-0933
SNS
Includes CD, Label & Jewel Case

**CD, CASSETTE, CDR and CD-ROM**
**Complete Packages!**
DDAI
DIGITAL DYNAMICS AUDIO INC
1-800-444-DDAI
www.4ddai.com
CD/CASS MASTERING
CD/CASS REPLICATION
CEDAR RESTORATION
GRAPHIC DESIGN
FILMS AND PRINTING
CD-ROM AUTHORING

**Complete CD packages!**
WAVE

| | CD's | Cassettes |
|---|---|---|
| 300 | $975 | $396 |
| 500 | $1075 | $506 |
| 1,000 | $1637 | $752 |

1-800-928-3310
World Audio Video Enterprises
Retail ready including full color inserts, Bar codes, CD Rom, CD R Mastering, Design and Film services available.

"OUR 30TH YEAR OF FAST GUARANTEED SERVICE"
FREE CATALOG
ANDOL AUDIO PRODUCTS INC.
4212 14TH AVENUE - BROOKLYN, NY. 11219
CD-RS, HI-8, MINI DISKS, S-VHS, MASTERING TAPE, A-DAT, R-DAT, BULK VHS, PLASTIC REELS, BOXES, NORELCO, SOFT POLY BOXES, VIDEO BOXES & SLEEVES, SINGLE & DOUBLE CD BOX, DCC, MOD, DDS2 AND MORE
CUSTOM LENGTHS FROM 126 MINUTES
100 PIECE MINIMUM

| LENGTH - TYPE | C-10 | C-20 | C-30 | C-40 | C-50 | C-60 | C-70 | C-80 | C-90 |
|---|---|---|---|---|---|---|---|---|---|
| BASF CHROME PLUS | .27 | .32 | .37 | .41 | .46 | .51 | .57 | .63 | .68 |
| TDK SA | .25 | .30 | .35 | .39 | .44 | .49 | .53 | .58 | .63 |
| MAXELL XLII | .27 | .32 | .38 | .44 | .47 | .52 | .67 | .70 | .76 |
| BASF CHROME SUPER | .30 | .38 | .47 | .56 | .65 | .69 | .80 | .87 | .89 |

TELEX · TDK · BASF · AMPEX · maxell
IN HOUSE DIRECT ON CASSETTE PRINTING
• CUSTOM PRINTED LABELS & J-CARDS •
IN NYC 718-435-7322 • 800-221-6578 • 24 HOUR FAX 718-853-2589
FREE LABELS WITH EVERY CASSETTE ORDER
ROUND EDGE OR STANDARD NORELCO BOXES $0.12 EACH
www.andolaudio.com
andolaudio@aol.com

ballistic.com
YOUR LINK FOR ALL COMPACT DISC, CD ROM & CASSETTE MANUFACTURING
$999.00 1000 CD'S
2 color on disc, insertion of your printed material, standard jewel box & polywrap. MANUFACTURED BY EMI
1.800.401.8273

**FAST CD-R DUPES!**
Lowest prices & Highest-quality work.
Sonic Solutions Digital Mastering
Major credit cards accepted.
Call or e-mail us for a free catalog.
DBS DIGITAL, HOBOKEN, NJ
(888) 222-2090
e-mail dbsdigital@aol.com

**EARTH DISC**
$986.00 500 CDs
package includes:
full color 2 page/4panel insert
3 color on cd printing/glass master,
jewel box, shrink wrap

$1299.00 500 CDs
we do it all
package includes:
graphic design/film
full color 2 page/4panel insert
3 color on cd printing/glass master,
jewel box, shrink wrap

1-800-876-5950

MASTERING ▫ MANUFACTURING ▫ PRINTING
**DIGI-ROM**
CD-AUDIO • CD-ROM
REAL TIME & HIGH SPEED AUDIO CASSETTES
COMPLETE PACKAGES • CD-ROM STRIKE-OFFS • GRAPHIC DESIGN STUDIO
CD-ROM PRODUCTION & ARCHIVING • *1 TO 200 DISCS DUPLICATED OVERNIGHT*
(800) 815-3444
(212) 730-2111
www.digirom.com
130 West 42nd Street • New York, NY 10036

**CD Packages!**
Includes: Glass Master, 3 Colors On Disc print, Jewel Case, Assembly & Overwrap!
300 - 599.00
500 - 749.00
1000 - 1090.00
800-579-7010
BASE Multimedia Inc. www.bmmi.com

**Musicians**
**Put Your Music On CD**
25 Full Color $8.00 ea
50 Full Color $6.50 ea.
100 Full Color $5.00 ea.
50 Free Cassettes with 100 CD Order
Call 1-800-446-4548
Lifetime Publishing-Paullina, IA

**Creative Disc**
& Web Services
Phone (615-356-4906) www.Creativedisc.com
Fax (615-353-1948) EMail Cds@home.com
CD REPLICATION
PRINTING
GRAPHIC DESIGN
BAR CODES
CASSETTE DUPLICATION
COLOR FILM SEPARATIONS
TYPESETTING
WEB SITES

1-800-TAPE WORLD or 1-800-245-6000
We'll beat any price! 6.95 SHIPPING • FREE CAT

| SONY | | MAXELL | | TDK | | FUJI | |
|---|---|---|---|---|---|---|---|
| DAT-120 | 5.49 | XL II-90 | 1.39 | CDR-74 NL | 1.49 | SVHS-T120 | 5.99 |
| DAT-124PRO | 8.99 | XL IIS-90 | 1.99 | CDRXG-74 | 5.79 | Hi 8 120 | 5.99 |
| T-120V | 1.69 | DAT-120 | 3.99 | SA-90 | 1.49 | DR-I-90 | .69 |
| L-750BT | 3.49 | DAT-124PRO | 6.99 | SAX-100 | 2.09 | JVCST120 | 3.99 |
| MDW-74 | 2.49 | T120HGX | 1.99 | T-120 EHG | 2.49 | JVC T-120 | 1.39 |
| SKC-CD-74 | .99 | T-160 GX | 1.99 | [illegible] | 8.99 | DR II-90 | .99 |

TAPE WORLD [illegible]

# CLASSIFIEDS

**Magnetic AIR**

INDIE BAND SPECIAL

300 CD's only $999.00

Package Includes: Consultation, full design services (incl. color proofs) and film negs. 2 Color CD with 2 page (1 panel) full color insert and full color tray card. Jewel case, shrink wrap and glass master.

We're putting you On Track

Visit our website at www.magneticair.com for other package options or call us at 1 800 561-1815 for a free brochure.

Visa/Mastercard accepted

---

**STANDARD MIDIFILES**

WHEN QUALITY COUNTS

ONLY **TRAN TRACKS** WILL DO

World's leader in quality & service
In business over 11 years
Over 4000 popular songs
Including Italian and Opera libraries
Rhythm, Groove and Style Disks
General MIDI compatible
Email service

**FREE demo & catalog**

**1-800-473-0797**

**www.trantracks.com**

TRAN TRACKS® MIDI SEQUENCES

350 Fifth Ave. #3304
NYC NY 10118
voice 973-383-6691
fax 973-383-0797

---

**100 CD's $349**
**1000 CD's $699**

EASTCO PRO Audio/Video Corp.

69¢ CD'S THEM BOYS IS CRAZY!

•Low, Low Quotes
•Guaranteed Quality

**1-800-365-8273**

•Complete Manufacturing: Audio & Video Dubs, DVD, CD's, Graphics, Printing & Packaging

---

www.recordinglair.com

RECORDING LAIR MASTERING

Russ Berger Design

Free Mastering Information

**888 881-LAIR**

---

Serving You — THE WAREHOUSE — Since 1975

Studio Sales & Services

Cassette, CD, CDR duplication
R-to-R · DATs · ADATs · Hi8 · VHS · CDs
Blank Cassettes, any length
Warehouse prices
800-483-TAPE · fax: 904-398-9683
Visa · MasterCard · Discover
internet: warehouse@jax.jaxnet.com
2071-20em Emerson St., Jacksonville, FL 32207 · 904-399-0424

---

**100 CDs** for only **$410!**

manufactured in as little as TWO DAYS

Includes
• FREE design
• FREE b/w insert & tray
• On-disc printing
• Jewel case

Other packages available.
Call for details and information packet.

(800) 249-1110 • (302) 999-1110

**National Multimedia Services**

Compact Discs • Real Time Cassettes
Digital Mastering

---

## SOFTWARE, SEQUENCES & SOUNDS

**DANGEROUS SOUNDS!** The best patches and samples for **Ensoniq keyboards**, from the ASR and TS back to the Mirage. Free Catalog! Syntaur Productions, (800) 334-1288, (409) 234-2700. Web http://www.fatsnake.com/syntaur

**WOW! PATCHMAN MUSIC 2 for 1 SALE** on all **SY77, TG77,** and **SY99** soundbanks! Take advantage of this unique opportunity to own PATCHMAN MUSIC quality. Hurry, offer ends 2/28/99! **For details call (216) 221-8282 or visit www.patchmanmusic.com!!**

**ENSONIQ OWNERS**: Convert Sequences to/from **Standard MIDI Files** on IBM/PCs. Each package TS-10/12, ASR-10, EPS/EPS-16, VFX-SD/SD-1, SQ-80, SQ-1/2, KS-32, or KT-76 costs $54.95. Convert SD-1 to TS-10 w/our SD1TS10 Conversion for $54.95. **Call for Alesis, Kawai, Korg, PianoDisc, Yamaha, Roland.** Visa/MC/Amex. **Giebler Enterprises**, 26 Crestview Drive, Phoenixville, PA 19460. **(610) 933-0332**; fax (610) 933-0395.

---

**Fleetwood MultiMedia**

1000 CDs from $799

500 Tapes Packaged $715

Includes Everything

Fast Turnaround

800-353-1830 (781) 599-2400

---

**10 YEAR ANNIVERSARY SALE!**

WE'RE SAYING THANK YOU BY GIVING YOU $10 OFF YOUR NEXT CD PURCHASE OR 10% OFF 3.5" DISKS

FUNKY BASS GUITARS

SPECIAL PRICE ONLY $89 + SHIP

OVER 1,000 DOPE GUIT. WHA. RIFFS. SOUNDS & LOOPS

You must mention this ad for sale prices

Also 3.5 disks for MPC, ASR, EPS.

(800) 301-6434 FAX 562 699-0864 WWW.MIDIMARK.COM

---

**Give Your Act A Good Kick In The Gas...!**

*Imagine what it would be like jamming with some of the best known musicians in the world. With Midi Hits, you just step in and play along.*

**Over 5,000 Premium Backing Tracks**

**Pop, Country, Oldies, Top 40, Big Bands, Jazz, Standards, Gospel, Show Tunes, Latin, R&B, etc.**

**Free Catalog**

Midi Hits

Now In our 7th Year

3 Shratton Ave.
San Carlos, CA 94070
**1-(800) 593-1228**
Fax (650) 637-9776
e-mail: midihits@pacbell.net

See us on the Web
**www.midi-hits.com**

---

WE'LL BEAT ANY PRICE ON CD'S

• CD
• CASSETTES
• HIGHEST QUALITY
• QUICK TURNAROUND
• PERSONALIZED SERVICE
• TOTAL COMPLETE PACKAGES
• LOWEST PRICES, CALL US LAST!

(813) 446-8273

**Total Tape Services**

639 Cleveland St / Clearwater, FL 34615

# CLASSIFIEDS

## SOFTWARE, SEQUENCES & SOUNDS

**PATCHMAN MUSIC's 2000-Series CD ROM** for K2000/2500 now contains 150 professional sound-banks! 3000+ Programs! Only $200. (Foreign $210). **To order, call (216) 221-8282 or visit www.patchmanmusic.com!!**

**K I D N E P R O**
**"The Patch King"**
*Quality Sounds for over 100 Midi & Vintage Instruments from:*
ROLAND • YAMAHA • ENSONIQ • KORG • EMU • AKAI • CASIO • KAWAI • KURZWEIL • OBERHEIM • SEQUENTIAL •

View our catalog & soundlists at our websight
http://www.kidnepro.com
Call for more info • Send $3.00 for catalog
**KID NEPRO PRODUCTIONS**
PO Box 360101 (DEPT E)
Brooklyn, NY 11236
(718) 642-7802 • Fax: (718) 642-8385
e-mail: kidnepro@aol.com

**TRYCHO TUNES**
**PERFORMANCE SEQUENCES™**
**MIDI MUSIC WITH THE HUMAN TOUCH!**
The most widely used midi sequences in the world. First in customer satisfaction for over twelve years. Over 5,000 song titles available for any musical application. All programmed in sunny California by Stephen Kern, the most respected musician in the business.
From Pop to Rock. From Country to Standards to custom programming, we've got it all!
Trycho sequences are available for most popular computer and hardware based sequencer systems. We even have stereo audio cassette and DAT versions for non-sequencer users. Now in our 13th year, we continue to offer great selection, great prices, and full time tech support. Just a phone call away six days a week!
Whatever your musical needs, you can count on ***TRYCHO TUNES*** for the absolute best in midi sequences.
***TRYCHO TUNES*** are available at many fine music & computer stores. Or order direct at:
**1-800-543-8988**
2166 W. Broadway St. • Suite 330
Anaheim, CA 92804
Technical Hotline (909) 696-5189 • Fax (909) 696-3571
http://www.trycho.com • email trycho@mindspring.com

**Got Samples?**


CD-ROM.................$199
*Check Out Our Full Line At*
**www.sonicreality.com**
**1-800-232-6186**

**SOUNDS!**
HUGE BULK SALE • WEB!: pro-rec.com
Info on all of our Sounds 24 hours a day!
SYNTH SOUNDS • CDs
CD-ROMs • SAMPLE DISKS
We're #1 in Synth Sounds in the World!
106 W 13th Street, Suite 13
New York, NY 10011 USA
Phone: (212) 675-5606
Fax & Demos: (212) 627-3148
email: sales@pro-rec.com
web! pro-rec.com
Pro-Rec

**BAND-IN-A-BOX IMPROVEMENT PRODUCTS*****You can put a Better-Band-In-Your-Box. Power-User Styles, Fake Disks & More! Gen-MIDI **SEQUENCE** & CD-ROMs, too! **FREE** info! Norton Music & Fun, Box 13149, Ft. Pierce, FL 34979. Voicemail/fax (561) 467-2420; http://members.aol.com/NortonMIDI/

***Music Tools Blowout!***
**Great Deals & Service**
**5th Anniversary Sale**
Software, Sound Cards, Interfaces, Cables, Controllers, Samples, Sequences, Books, Videos
**Shop for over 12,000 products at**
**www.midi-classics.com**
**Call 800-787-6434 NOW!**
MIDI Classics, Dept.E, Box 311, Weatogue, CT 06089

***TIRED OF WAITING ?***
CONVERT YOUR LIST OF EDITED MONO FILES TO STEREO WAV FILES FOR RECORDING YOUR CD
**FAST AND EASY !**
MonoSter 1.0 converts your mono cue file lists using super-fast algorithms working in the background. " Hands Off " conversion of your entire list of mono files automatically !
MonoSter 1.0 for PC. DOS / Win 3.11 or Win 95 / 98
Segue data Systems™
**Only $ 29.95 ! ! !**
Call to Order (818) 558-1222 or order from the internet www.seguedata.com

**WORLD CLASS MIDI FILES**
**the WORKS Music Productions**
For **Free Catalog & Demo Disk** call **(800) 531-5868** or visit our web site: **www.worksmidi.com**
Popular styles, General MIDI compatible, e-mail delivery avail. Box 22681, Milwaukie, OR 97269.

**THE BEST MIDI SEQUENCES MONEY CAN BUY**
**Classic Rock, R&B, Blues, and Jazz standards** programmed by **Pete Solley**.
**LET US SEND YOU OUR FREE DEMO DISK AND SEE WHY WE SIMPLY ARE THE BEST.**
Call (888) 211-0634
or fax (954) 570-9788 for song list.
**CHECK OUT OUR NEW STYLE DISKS**
All credit cards accepted.
Visit our Web site at
www.petersolleyproductions.com
**Peter Solley Productions**

**MPC 3000 & 2000 USERS (888)-349-4488!!**
**e-mail vinyl@optonline.net**
**For $299 you get a Zip Disk with 26 Programs with nothing but *Raw* samples (All from vinyl - no sound CD's). Individually chopped up into the PADS. Samples contain all types of HITS, RIFFS, PERC, and a whole lot of miscellaneous stuff. If you order now, you can get both VER 1.0 (26 programs), & VER 2.0 (with over 40 programs) for only $399!**
**Remember we don't sample off CDs. Everything is strictly *VINYL!***

**Translator™!**
**Say goodbye to proprietary formats! Read ANY sample format CD-ROM or SCSI Drive for access to Emu®, Roland®, Kurzweil®, Akai®, or Ensoniq® samplers!**
Translate absolutely any popular disk or file format (all samplers, .WAV, .AIF, more) with great results. Translate ALL parameters, keymaps, samples - absolutely EVERYTHING!
- Write native format SCSI Drives
- Transfer via SCSI (SMDI or other)
- Edit any parameter onscreen

**$149.95!**
**Chicken Systems**
www.chickensys.com/translator
800.877.6377 - 320.235.9798

**MIDI HQ**—Your High-Quality HeadQuarters for Standard MIDI Files & More! Over 2,500 Licensed hits! For a free catalog call: (800) 844-4785; outside US, (843) 293-3767. www.midihq.com

*Color Classifieds in Electronic Musician Get Results!*
Reach over 62,000+ potential buyers for your product & service with a color, eye-catching classified ad in EM. Classifieds in EM are the most cost-effective way to get visibility for your company month after month!
For Classified advertising rates in Electronic Musician, call Robin Boyce-Trubitt at (800) 544-5530 or Email: emclass@intertec.com

# CLASSIFIEDS

## Learn *AND* Play Your Favorite Songs Now!

**CD Looper** is the perfect music software for Windows that allows you to easily learn how to play any song directly from your computer's cd player. With CD Looper's ability to slow down any audio cd **2**, **3** or **4** times **without** changing pitch, you can easily learn to play your favorite songs note for note!

- Set unlimited loops with 1/100th of a second resolution
- Loops can be any length
- Sped up loops in 10% increments
- Many other features

**$59.99**

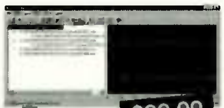
### CD Looper Pro Includes These PlugIns!

**NoteGrabber:** Extends CD Looper's capabilities beyond CDs. Record and slow down music from any input source or wav file. Graphically create loops down to a single note!
**OverDubber:** Record yourself playing over any loop or wav file in NoteGrabber. Record yourself playing over backing tracks.
**PitchChanger:** Change loop's pitch in half step increments.

CD Looper Pro is **$94.99** or you can upgrade from CD Looper for only **$39.99**

**DeComposer** is an advanced filtering program for Windows that easily removes any instrument from digital audio files.

### Features



- Exclude note ranges and create your own backing band.
- Include note ranges and focus on only the instrument you want to hear.
- Select from many preset instrument ranges.
- Easily create LowPass, HighPass, BandPass, BandStop and Notch filters.
- Boost or cut the volume of any frequency range.
- Remove 50/60 hertz electrical noise & high-pitched hiss.

**$99.99**

** "We found both programs very useful for studying a specific artist or material in depth"
Guitar World - 10/98

Order both CD Looper Pro and DeComposer and **save!**

CD Looper

Order today! Call RePlay Technologies toll-free at **(888) 3RePlay** or place your order directly on our web site at **www.replayinc.com**. Next day delivery available.

DeComposer

**COMPUTER MUSIC PRODUCTS**
Sound card MIDI adapter cables, MIDI & digital audio software, hardware & accessories. Online catalog:
**www.musicmall.com/cmp**
MIDI Tutorials: **www.jjonline.com/digital/miditutorial.asp**

## SOUND EFFECTS

**Sound Effects That Scream!**
CONCEPT:FX, 195 AIFF and WAV sounds, royalty-free.
Mac/PC CD-ROM.
$49.95 + $4 shipping.
**F7 Sound and Vision**
17732 Nathan's Drive
Tampa, FL 33647. **(813) 991-4117**
http://www.f7sound.com

## EM CLASSIFIEDS WORK FOR YOU

| | |
|---|---|
| **Text rate:** | **$9.75 per line (approximately 25-32 character spaces per line); seven-line minimum. Add $0.50 per bold word. Each space and punctuation mark counts as a character. $68.25 MINIMUM CHARGE for each ad placed.** |
| **Enhancements:** | $50 per inch 4 color, $10 black border, $25 for one color screened background, $25 for a reverse. $25 for Post Office box service. Charges are based on a per-insertion basis. |
| **Display rate:** | $125 per inch (1" minimum/half-page maximum). Logos or display advertising must be camera-ready, sized to **EM** column widths and specs. Frequency discount rates available; call for information. |
| **Special Saver rate:** | $35 for up to four lines, including first word in bold. **Only available to individuals not engaged in commercial enterprises.** No additional copy allowable for this rate. |
| **Closing:** | **First of the month**, two months preceding the cover date (for example, the April issue closing is February 1). Ads received after closing will be held for the next month unless otherwise stated. **Cancellations will not be accepted after the closing date**. Copy changes and cancellations must be submitted in writing. |
| **Other requirements:** | Full street address (PO boxes aren't sufficient) and phone number must accompany all requests, whether included in ad or not. All words to be **bold** should be underlined. Copy must be typed or printed legibly in standard upper/lower case. Publishers are not responsible for errors due to poor copy. Arrangement of characters may be altered in typesetting process due to space. The publishers are not liable for the contents of advertisements. |
| **The small print:** | Only ads dealing with music, computers, or electronics will be accepted. No stated or implied discounts allowed on new-equipment sales. Publishers reserve the right to refuse or discontinue any ad deemed inappropriate. |
| **Art Instructions:** | Logos or display advertising must be sized to **EM** column widths and specs. For best printing results please provide exact size film (emulsion side down) preferably with a velox proof, or camera-ready linotronic paper output, or a stat. We accept laser prints or photo copies but do not assume responsibility for their reproduction quality. Line screen should be between 90 & 133 LPI. |
| **Send coupon & payment to:** | **Electronic Musician Classifieds:** Attn: Robin Boyce-Trubitt, 6400 Hollis St., #12, Emeryville, CA 94608 tel. **(800) 544-5530** or (510) 653-3307; fax (510) 653-8171; e-mail emclass@intertec.com. |
| **Payment:** | **Must be included with copy: check, Visa, MasterCard, or American Express accepted. Sorry, no billing or credit available.** |

INSERT THIS AD IN THE ________ ISSUE OF EM
Categories available (PLEASE CHECK ONE):
- ☐ ACOUSTIC PRODUCTS
- ☐ COMPUTER SYSTEMS
- ☐ EMPLOYMENT OFFERED
- ☐ EQUIPMENT FOR SALE
- ☐ EQUIPMENT WANTED
- ☐ INSTRUCTION
- ☐ INTERNET SERVICES
- ☐ MAINTENANCE SERVICES
- ☐ PARTS & ACCESSORIES
- ☐ RECORDING SERVICES & REPAIRS
- ☐ RECORDS, TAPES & CDS
- ☐ SOFTWARE, SEQUENCES & SOUNDS
- ☐ SONGWRITING
- ☐ SOUND EFFECTS

ATTACH YOUR CLASSIFIED AD COPY ON A SEPARATE SHEET, TYPED DOUBLE-SPACED OR PRINTED CLEARLY IN CAPITAL AND LOWER-CASE LETTERS.

**Company Name** ________
**Name** ________
**Address** (no PO boxes) ________
**City** ________
**State** ________
**Zip** ________
**Phone** (   ) ________
**Signature** ________

| | | |
|---|---|---|
| ________ | Display ($125 per inch) | $________ |
| ________ | Lines @ $9.75 (seven-line minimum) | $________ |
| ________ | Bold @ $0.50 additional | $________ |
| ________ | Border @ $10 | $________ |
| ________ | Reverse @ $25 | $________ |
| ________ | Color Screen @ $25 | $________ |
| ________ | 4-Color @ $50 per inch | $________ |
| ________ | Special Saver rate = | $ 35 |

TOTAL PAYMENT INCLUDED $________

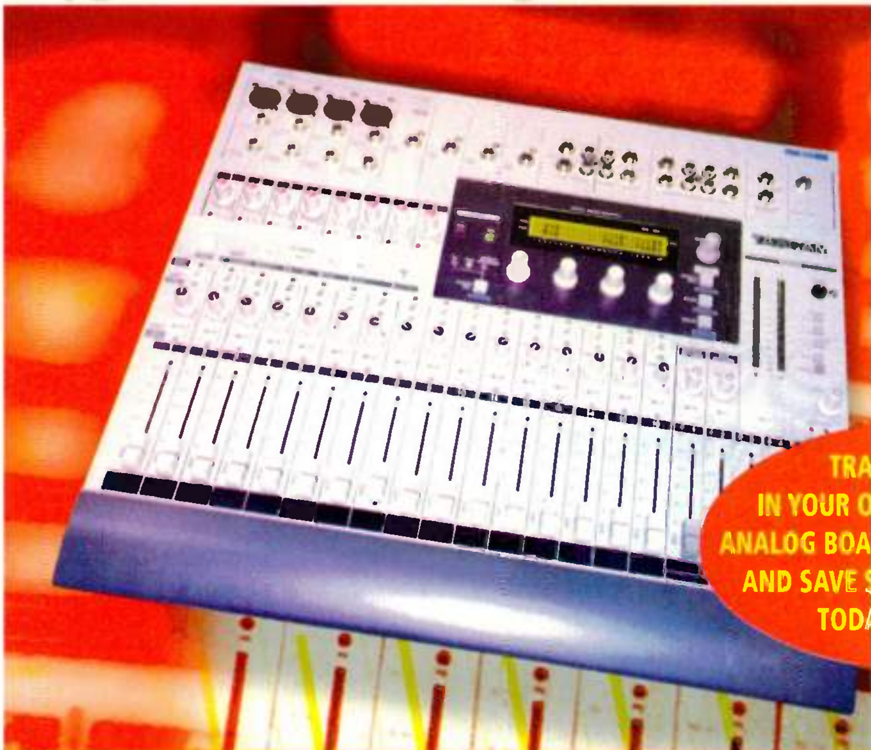
☐ Visa ☐ MC ☐ AmEx ☐ Discover

☐ Check/Money Order # ________

Card # ________ Exp. ________

## Why compromise your sound? Upgrade to TASCAM Digital today!

# A digital 8-track with an analog board?




- 8 Analog Inputs
- XLR Mic Pres with Phantom Power
- 8 Channels of Digital I/O
- AES/EBU and S/PDIF Digital I/O
- 16 Channel Strips with Faders
- 128 Scene Snapshot Automation
- On-board Dynamics Processing
- 10 Stereo Effects
- 3-Band Sweep EQ
- 4 Group/Aux Buss and Stereo Master Fader
- 8 Channel Monitor Section
- MMC and Word Clock Synchronization

So much more, we've run out of room! Call for more info!

OK, we understand. The price/performance revolution hit digital recorders first with ground-breaking products like the DA-88, DA-38 and ADAT®. So you got yourself a great digital multitrack. But making the jump to a digital board was another story — a delicious idea, but those $10,000 desks were just too big a stretch. Still, in your heart, you knew there had to be a better way.

Well, thanks to the folks at TASCAM, now there is — the stunning TM-D1000 digital mixer, designed for both your digital multitrack and your budget! With a mind-bending feature set and crystal-clear sound quality, the TM-D1000 is the board you've longed for ever since you first got your digital multitrack. Taking it over the top is the unbelievable list price — only $1,299! TASCAM has gone all out on this one, providing maximum value and making your dream of an all-digital studio a reality.

The clincher on why to act now? Sweetwater Sound will pay you top dollar in trade for your old analog console when you upgrade to a new TASCAM TM-D1000! No lost time and money running want ads and having strangers poking around your studio. We'll even send you a shipping box and issue a pick-up call if need be — what could be easier?

**Call now to trade in your old board. Build your all-digital studio with your new TASCAM TM-D1000 Digital Mixer!**

# (800) 222-4700

**Sweetwater SOUND INC.**

5335 Bass Road, Fort Wayne, IN 46808
(219) 432-8176 · FAX (219) 432-1758
www.sweetwater.com · sales@sweetwater.com

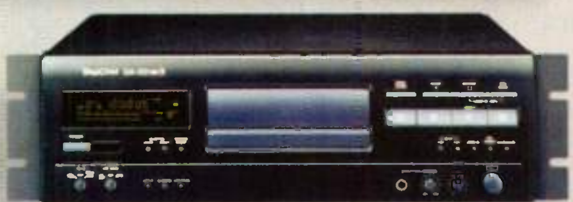
SEE US AT NAMM BOOTH #1827

**MUSIC TECHNOLOGY DIRECT & THE BEST VALUE GUARANTEED!**

# Upgrade your DAT to TASCAM 24-bit!

***Sweetwater pays you top dollar in trade when you upgrade your stereo mastering deck to a* TASCAM *DAT recorder! Call now for details!***


*DA-302 Dual DAT Master*



*DA 20MKII Project Studio DAT Master*

*DA-30MKII The Studio DAT Master*

*DA-P1 Portable DAT Recorder*

## The World's First 24-bit DAT Recorder is Here. The TASCAM DA-45HR (No Hype Required).

That's right! Now you can have 24-bit mastering for your music — without a huge computer system. The brand new **DA-45HR** advances the state of the art for DAT recorders to genuine 24-bit quality. The DA-45HR connects to today's top 24-bit mastering processors with a single cable. Plus, it's the most economical DAT machine in the world that includes Word Clock.

You want your precious masters to sound great no matter what. 24-bit is the wave of the future — and it's yours today from **TASCAM** and **Sweetwater Sound!**

In fact, TASCAM has a complete family of powerhouse DAT recorders. Consider another TASCAM worldwide first — the dual DAT **DA-302**. Perfect for fast dubbing and realtime safeties, including 24-bit DATs. Everyone knows the industry standard **DA-30MKII** — it's still the price-to-performance leader. Going remote? The **DA-P1** portable DAT has great mic preamps and all the pro features needed for location recording and sampling. And the **DA-20MKII** provides bold TASCAM quality at a very modest price, making it perfect for the personal studio.

So, pick your TASCAM DAT. Then Sweetwater will give you top dollar in trade for your old mastering recorder. It simply couldn't be any easier or more affordable to upgrade to the DAT of your dreams!

**Why not call right now and find out just how easy and economical it is to get your dream DAT from TASCAM?**

**(800) 222-4700**

Sweetwater SOUND INC.
Music Technology Direct & the Best Value Guaranteed

5335 Bass Road
Fort Wayne, IN 46808
(219) 432-8176
FAX (219) 432-1758
sales@sweetwater.com
www.sweetwater.com

**SEE US AT NAMM BOOTH #1827**

circle #638 on reader service card

# The New Dream Team

## For Your PC or Mac

## BlueMAX

### Smart Compressor with 15 Presets

"look past the BlueMAX's small price and diminutive package, you will see a box that does what no other analog compressor has ever done before and sounds great as well." - George Peterson, MIX Magazine. Finally - presets on a compressor!

## ACP88

### Eight Channel Compressor/Limiter/Gate

Includes Manual & Automatic Compression Curves, Full Metering, Selectable Gate Range, Side Chain and Gate Key on every channel. Brick Wall Limiting for in-ear monitors and maximum resolution digital recording. Yes, just two rack spaces!

Electronic Musician 1999 EDITORS CHOICE

## ACP22

### Dual Channel Compressor/Limiter/Gate

Unique features include Program-Dependant Auto Mode, Hard/Soft Knee, Selectable Gate Range, Independent Gate Key & Sidechain and a Lo Pass Gate Filter that stops high frequencies (cymbals, etc.) from triggering the gate without affecting the audio. Exacting control in stereo!

## MP20

### Dual Channel Microphone Preamplifier/DI with Mix Bus

Two tasty mic pres with Jensen® Transformers, FET Class A Discrete Buffers. Twin Servo® Gain Stages (no capacitors) and High Headroom Mix Bus. Unique IDSS Control adjusts "warmth" on each channel. Convenient front-panel instrument inputs for DI use. The perfect gift for your favorite pair of microphones!

## M-80

### Eight Channel Microphone Preamplifier/DI with Mix Bus

Eight delicious mic pres with Jensen® Transformers, FET Class A Discrete Buffers. Twin Servo® Gain Stages (no capacitors) and High Headroom Mix Bus. Unique IDSS Control adjusts "warmth" on each channel. Each channel also has 1/4" input for instrument DI. Record direct to your HDR or MDM rig!

# Feel The

PreSonus products have rapidly become favorites with the many musicians and engineers here at Sweetwater Sound. They deliver stunning sound and innovative features while shattering price/performance barriers. From their "overnight sensation" BlueMAX compressor to the amazing M80 eight channel mic preamp with Jensen® transformers, PreSonus brings you unparalleled value.

Sweetwater SOUND INC.

*Why not call Sweetwater Sound right now and find out how PreSonus gear can enhance your recordings? You could be "feeling the vibe" tomorrow*

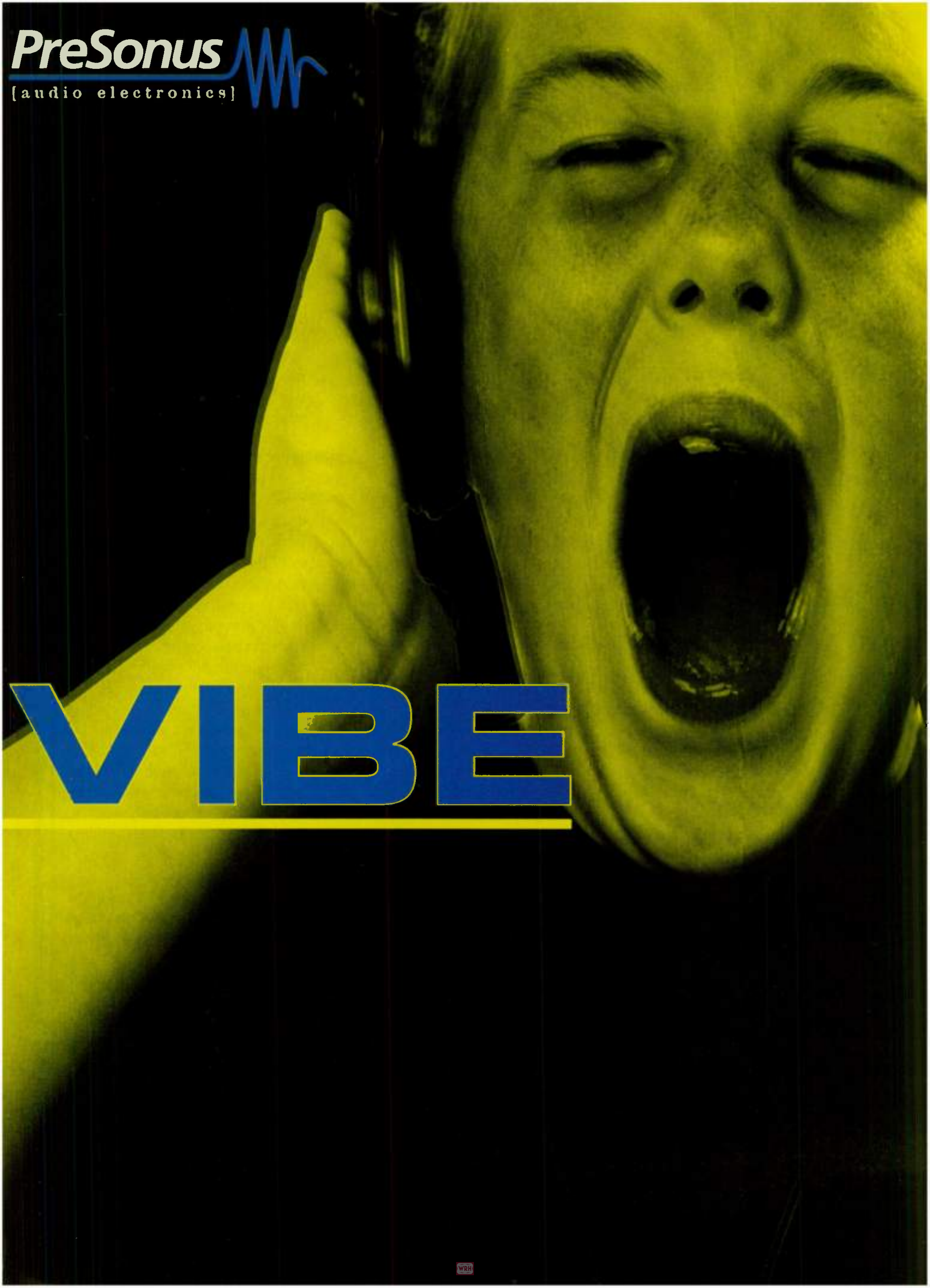
# CALL (800) 222-4700

circle #641 on reader service card

SEE US AT NAMM BOOTH #1827

*Music Technology Direct - and the Best Value, Guaranteed.*

www.sweetwater.com • sales@sweetwater.com • (219) 432-8176 • FAX (219) 432-1758 • 5335 Bass Road • Fort Wayne, Indiana 468



PreSonus
[audio electronics]
VIBE

# FINAL MIX

By Larry the O

## A Model Citizen

For years, we've all been hearing the buzzword *physical modeling.* Emulating acoustic instruments by creating mathematical models of their physical systems holds out the tantalizing possibility of musicians being able to play electronic musical instruments as expressively as they can acoustic instruments. Unfortunately, as the song says, "It don't come easy." Significant obstacles to realizing that potential exist in all directions.

The first obstacle—the required computing power—should soon be overcome. Personal computer CPUs have increased hugely in power and speed since their introduction, and I'd guess ASICs (custom chips) for modeling aren't far behind.

The second challenge is finding ways to control the models. What makes acoustic instruments so expressive is the sophistication of control that is achieved through the manipulation of a relatively small number of mechanical systems. Each system typically offers a number of parameters that can be varied interactively, resulting in a spectrum of effects. In combination, these systems yield a universe of expression. Take the vibrating reed of a clarinet: lip pressure, air pressure (velocity and volume), and wetness are just three parameters a player can manipulate with the mouth and lungs alone. Other parameters, such as reed thickness, can be varied out of real time.

To achieve such a level of expression out of a MIDI controller with a method as efficient as a reed and mouthpiece between your lips is difficult, if even possible. One needs a physical controller that allows sufficient nuance, as well as software that takes the input from the controller and translates it into parameters of the model.

The third challenge is building models. Currently, making a physical model is a long, involved process that requires great expertise and powerful tools. Each model is essentially hand built and endlessly tweaked. The laboriousness of the process means that few models are available to run on the commonly available modeling engines. Modeling will become more powerful and ubiquitous when making a model is as uncomplicated as making a multisample is today.

Imagine a day when you can take a handful of recordings of the sound you want to model, drag and drop them on a personal computer application, and come back later to a new file for your modeling engine. (You'll always have to do some tweaking to get things just right, but that's perfectly reasonable.) Such a day will come, but it sure as heck won't be next week.

Finally, modeling can be used in many more ways than just creating musical instruments, as exciting and wonderful as that is. In the world of computer games, there has been a lot of talk about using modeling in place of the FM wavetable synths currently found on sound cards. This idea presumes that the music is composed and delivered as MIDI files and that it has merit. But the same increasing power in CPUs that has made the use of modeling feasible also makes it easier to play digital audio. It seems easier and, at present, cheaper to bring in some good players for a session than to spend the time trying to tweak models—in addition to which, real players sound better!

On the other hand, think about the interactive sound effects in a game. Say, for example, you have a game that is all about sword fighting. If you had models of sword swings and hits, every swing or hit could sound different throughout the whole game, as well as every time you play it.

I think modeling will be one of the standard ways of synthesizing sound in the future. However, it's many a mile down the Yellow Brick Road before we reach that Emerald City of sound.

---

**Larry the O** *is a musician, producer, engineer, and sound designer whose San Francisco–based company, Toys in the Attic, provides a variety of musical and audio services. He does not have a TV, VCR, cell phone, or Web page, but he holds the distinction of introducing the term* stud muffin *to audio writing.*

ILLUSTRATION BY STEVE CURL

# Sample Accurate Timing

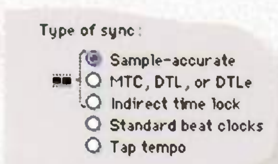
*Why settle for anything less?*

## Edit at the sample level in Digital Performer's multi-track audio editor

Feel the confidence of having total control over your audio – down to the very sample. Zoom all the way in to the sample level so you can see exactly what's going on in your tracks. Trim a region visually to the nearest zero crossing. Nudge your audio one sample at a time. Got phase problems between two or more tracks? Fix them. Graphically. Need to tweak a few samples here and there? Just redraw them with the pencil tool in the new stereo waveform editor. Digital Performer gives you all the sample-level editing you need.


## Make Digital Performer a sample-accurate extension of your ADAT or Tascam tape deck

Digitally transfer tracks into Digital Performer via our 2408 audio interface, and say good-bye to worrisome phase issues and time-consuming sync workarounds. Thanks to ADAT sync (and our Digital Timepiece Control Track Sync for Tascam), your audio won't drift a single sample - even if you fly an audio region back and forth a dozen times. Other popular (and costly) workstations introduce drift with every transfer. Why settle for anything less than perfect timing? With Digital Performer, the 2408 does it for under $1,000.




*Does your audio software support sample-accurate sync?*

## New for Digital Performer 2.5

- Built-in stereo waveform editor — you won't need to spend extra hundreds on separate waveform editing software.
- Sample-accurate editing — view and edit audio at the sample level in Digital Performer's multitrack audio editor.
- MasterWorks mastering plug-ins — 64-bit multiband compressor and limiter.
- Digidesign Direct I/O support — run Digital Performer's native real-time effects on any current Digidesign system including Project II and ProTools | 24 MIXplus.
- Plug-in enhancements — Apply any real-time MAS plug-in as a region operation.
- Support for popular audio I/O cards such as the Sonorus STUDI/O, Yamaha DSP Factory and Event Gina, Darla & Layla.
- Graphic parametric EQ — tweak up to 8 bands visually with floating numericals. Simultaneously drag multiple filters.
- Insert Measures — add time to the beginning of a project while preserving all subsequent material, including tempos and hit points.
- Navigation Enhancements — get around more efficiently than ever.
- Project copying and backup enhancements — easily "collect" all related audio/fade/analysis files when copying and archiving projects.


*The MasterWorks multiband compressor and limiter plug-ins put a mastering studio inside your Macintosh... and they're **included** with Digital Performer.*

Mark of the Unicorn, Inc. • 1280 Massachusetts Avenue • Cambridge MA 02138 • 617-576-2760 • 617-576-3609 fax • www.motu.com



circle #501 on reader service card