SYNTH-IN-A-BOX: SOFTWARE SYNTHESIS FOR THE DESKTOP STUDIO

Electronic Musician

October 1995

POWER TOOLS

4 power amps flex their sonic muscles

Record breathtaking vocal tracks

Promote your act with e-mail

Super Producer Tony Visconti

IT'S UNLEASHED MORE CREATIVITY THAN

The Mackie CR-1604 16x2 mic/line mixer is getting dangerously close to becoming a pro audio classic.

Not because it has the most mic inputs, knobs, buses or switches — others have long since topped us for sheer numbers of doo-dads and thingamabobs.

No, the CR-1604 has ended up in so many studios and on so many stages because it sounds good. And because it's downright easy to use.

More than any other component, your mixer is the focal point of all your creative efforts. If it's complicated to operate, you've just erected a frustrating barrier between you and your music. If it's noisy, everything that's recorded will be noisy, too.

When you're looking for an affordable, compact mixer that's good enough to regularly record complete albums and primetime TV soundtracks, call us toll-free (M-F 8:00 AM to 5PM PT).

You'll talk to a real person who'll send you our 40-page color tabloid complete with a 16-page hook-up and applications guide.

Then start exercising your musical creativity with the mixer that's becoming a classic for all the right reasons.

EXTREMELY RUDE, BLINKING SOLO LIGHT. *Sounds like a minor detail until some night at 2AM when you can't figure out why there's no sound coming out of your monitors.*

BEEFY HEADPHONE AMP WITH SEPARATE VOLUME FADER. *Instead of the usual wimpy amp, the CR-1604 has a separate, high-gain headphone amplifier section with enough gain to drive any brand of headphone to shock volume levels that will satisfy even a drummer. Also has more than enough gain to drive any monitor amplifier.*

INSIDE: QUALITY COMPONENTS *like double-sided, through-hole plated fiberglass circuit boards with solid brass stand-offs, gold-plated interconnects and sealed rotary potentiometers that resist dust & liquid contamination.*

BEST RFI PROTECTION OF ANY COMPACT MIXER. *No matter how quiet a mixer's internal circuitry is, it can be sabotoged by external radio frequency interference. RFI is created by broadcast stations, cell phones, computers and even that expensive radio-controlled car your kid got for Christmas. RFI gets into a mixer via the input jacks where it uses the internal circuit traces as miniature antennas to produce noise ranging from a low-level hiss to actual, audible voices and music.*

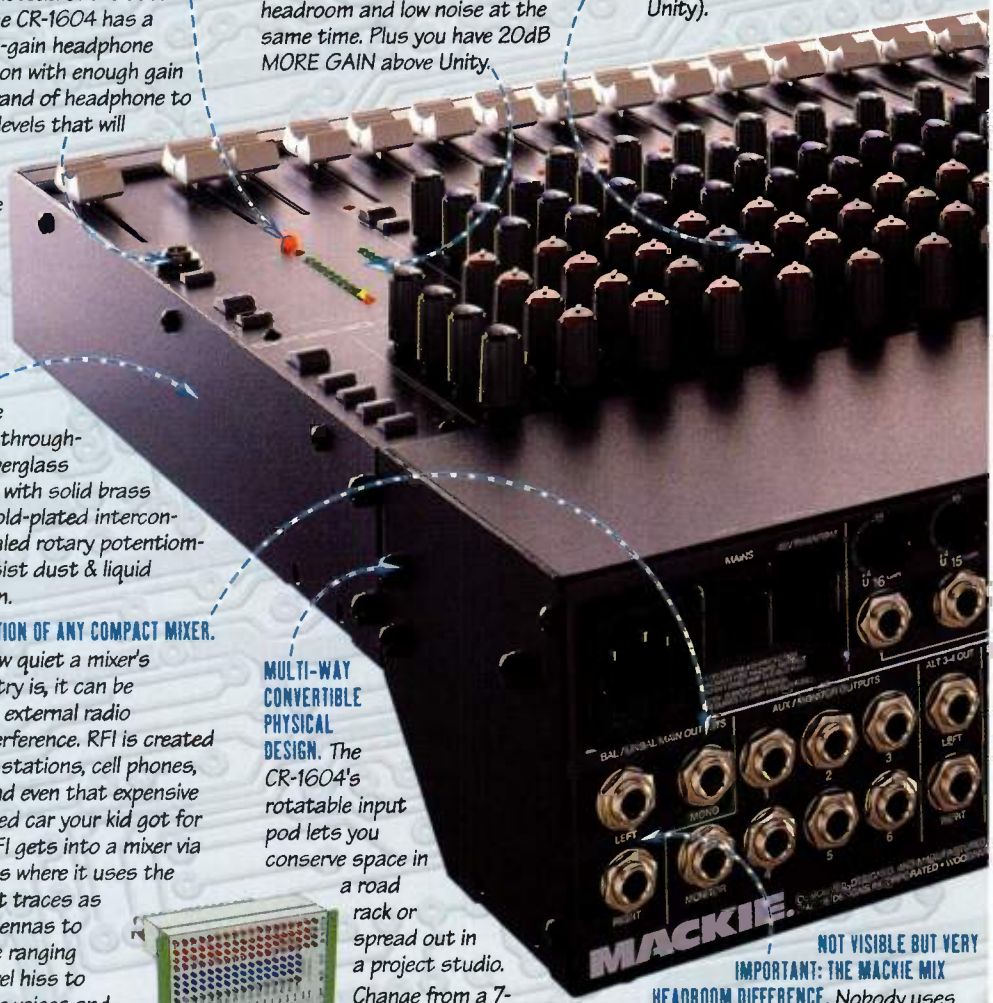
The CR-1604's 1/4" jacks use a shunting capacitor to stop RFI before the main circuit traces. Instead, RFI is re-routed back through the metal jack body and washer, then dissipated via the mixer's outer chassis.

XLR inputs are likewise protected from RFI via ferrite beads.

Next time you see a mixer with plastic 1/4" jacks, remember what you just learned.

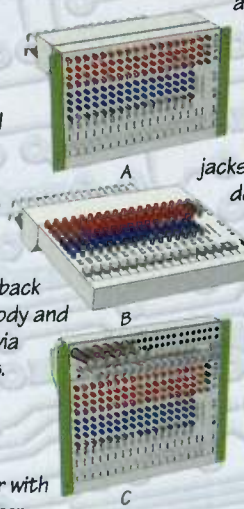
DUAL PURPOSE METERING SYSTEM. *Besides showing main L/R output level, the LED ladders are used to establish input levels. Set a channel fader at Unity, press the channel's SOLO button and set input trim level. This approach achieves very high headroom and low noise at the same time. Plus you have 20dB MORE GAIN above Unity.*

INSTANT HANDS-ON-ACCESS *to constant power pan controls, musical 3-band equalization, ALT 3/4 extra stereo bus, stereo in-place solo, seven high gain Aux sends per channel (via four controls) and four high gain stereo Aux returns (20dB more gain above Unity).*



MULTI-WAY CONVERTIBLE PHYSICAL DESIGN. *The CR-1604's rotatable input pod lets you conserve space in a road rack or spread out in a project studio. Change from a 7-rack-space mixer with jacks to back (A) to a tabletop design with jacks to top (B) in minutes. Add our optional RotoPod bracket (C) and rotate inputs and outputs to the same plane as the mixer's controls (a favorite for small SR set-ups).*

Any of these conversions takes just minutes with a Phillips-head screwdriver. And our XLR10 10-mic-preamp expander can be added in any of the configurations.



NOT VISIBLE BUT VERY IMPORTANT: THE MACKIE MIX HEADROOM DIFFERENCE. *Nobody uses just one channel of a mixer(although most headroom specs are stated that way). In any mixer, the mix amp stage combines signals from ALL inputs at once. If it overloads, you can't back off the master fader because it comes AFTER the the mix amp. So audible distortion results when the mix amp gets bogged down with multiple hot inputs. Mackie's unique mix amp architecture provides as much as twice the mix headroom of conventional designs. No wonder it's a favorite of top electronic percussionists.*

ANY OTHER COMPACT MIXER.

ULTRA-LOW NOISE. *When you compare noise specs, look for the one that counts: all 16 channels up at Unity Gain — not one channel at Unity gain. No other compact mixer beats the CR-1604 when it comes to low noise floor.*

MULTI-FUNCTION AUX SEND SYSTEM WITH LOADS OF GAIN. *AUX 1 on each channel can be used either for effects (post-fader/pre-EQ) or switched to monitor sends for stage monitor or headphone cue signal (pre-fader/pre-EQ). AUXs 2 thru 6 are post-fader/post EQ. AUX 3 and 4 knobs can be shifted to AUX 5 and 6 at the touch of a button.*

MUSICAL 3-BAND EQUALIZATION. *The CR-1604 redefined equalization points for compact mixers: 12kHz Hi EQ (instead of 10kHz) for more sizzle and less aural fatigue, 2.5kHz Mid (vs. 1kHz) for better control of vocals and instrumental harmonics, and 80Hz Lo EQ (instead of 100Hz.) for more depth and less "bonk." Others have copied these EQ points, but none have successfully emulated our quality equalization circuitry. It costs us more, but the result is zero phase distortion and a sweeter, more musical sound. It's another reason that the CR-1604 is a favorite of TV and film soundtrack scorers.*

LEGENDARY MICROPHONE PREAMPS. *Instead of sixteen "acceptable" integrated circuit microphone preamps, the CR-1604 features six big-console-quality preamps...the same mic preamp design that's on our acclaimed 8•Bus consoles. You get tremendous headroom and bandwidth with less noise and distortion. If your particular application requires more mic inputs, simply add our XLR10 10-Mic-Preamp Expander. Both it and the CR-1604's internal mic preamps have real and verifiable specs of –129.5 dBm E.I.N., 300,000Hz bandwidth and 0.005% THD. No wonder several of the world's top microphone manufacturers use Mackie Designs CR-1604s to demo their finest condenser mics at trade shows.*

INPUTS AND OUTPUTS AT PROFESSIONAL — NOT HOBBYIST — SIGNAL LEVELS. *The CR-1604 operates internally at industry-standard +4dBu levels to help reduce noise. But it can also handle the weaker –10dBV levels found on some digital multitrack machines and other equipment.*

THE PERFECT MATCH FOR ADATs, DA-88s AND HARD DISK RECORDING SYSTEMS. *We'd dearly love for you to buy one of our 8•Bus in-line consoles, but the CR-1604 makes a very effective 8-track recording mixer. The CR-1604's first eight channels have post-fader channel inserts (channel access). This VERY important feature is found on few other compact mixers. It lets you create a "split console" so that you can simultaneously track on eight channels and monitor/mixdown on eight more.*

EXPANDABILITY. *If you add a second or third digital multitrack, you can use one or two additional CR-1604s with our MixerMixer active combiner. It lets you run 32 or 48 channels without having to "cascade" the mixers.*

[1]This is no idle boast. Consider these tours for starters: Madonna, Rolling Stones, BoyzIIMen, Whitney Houston, INXS, Janet Jackson, Peter Gabriel, Bette Midler, Bruce Springsteen, Paula Abdul and Moody Blues. Mention in this list denotes useage by band members or tour techs and in no way constitutes an endorsement by the artists mentioned.

[2]More fine print: Mention in this ad denotes useage as reported to Mackie Designs and in no way denotes endorsement by the artist, program or production company listed.

- Split monitor configurable for easy 8-track digital tracking & mixdown
- Used on more superstar world tours in the last three years than all other compact mixer brands combined[1]
- Legendary studio-quality discrete microphone preamps
- Used by members of the Tonight Show band, David Letterman band, Conan O'Brien band, Saturday Night Live Band[2]
- Expandable with XLR10 Mic Preamp Expander & Mixer Mixer active combiner
- Used by Fox Television Sports for Monday Night Football on-field sound[2]
- Built-like-a-tank physical construction (it's too darned homely to be fragile)
- Used for sound design and incidental musical scoring on the world's most popular TV show
- Special mix amp architecture for twice the mixdown headroom of other designs

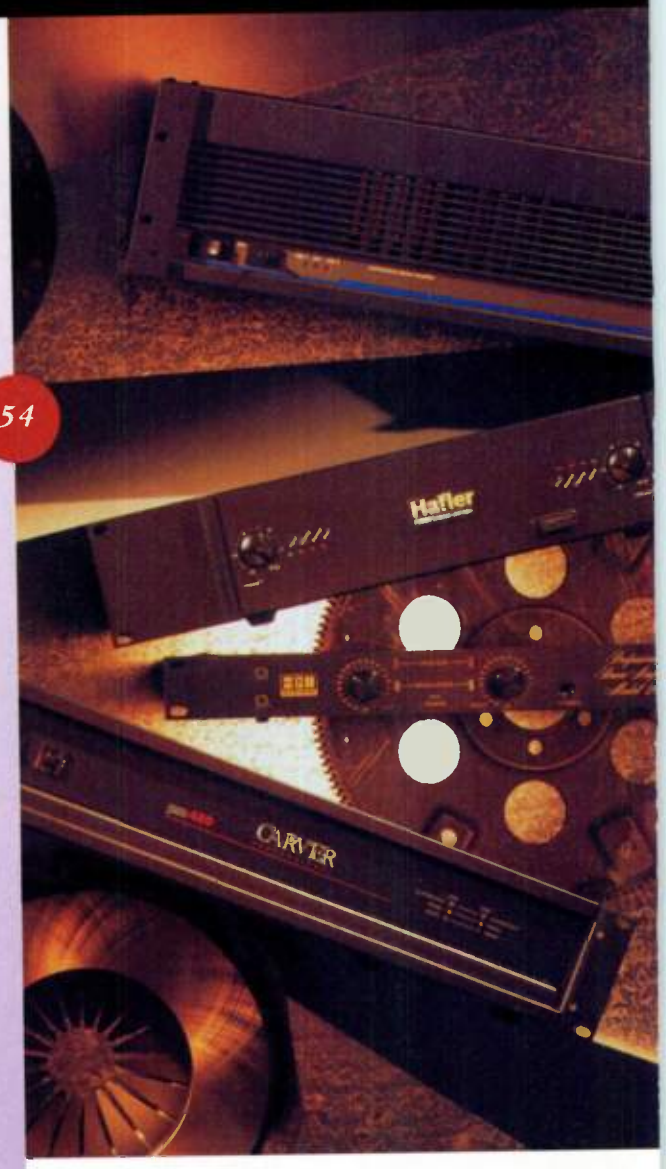
MACKIE™

16220 Wood-Red Road NE • Woodinville • WA • 98072 800/898-3211 206/487-4337
mackie.com • Outside the US, 206/487-4333 • Represented in Canada by S.F. Marketing 800/363-8855

WRH

I N S I

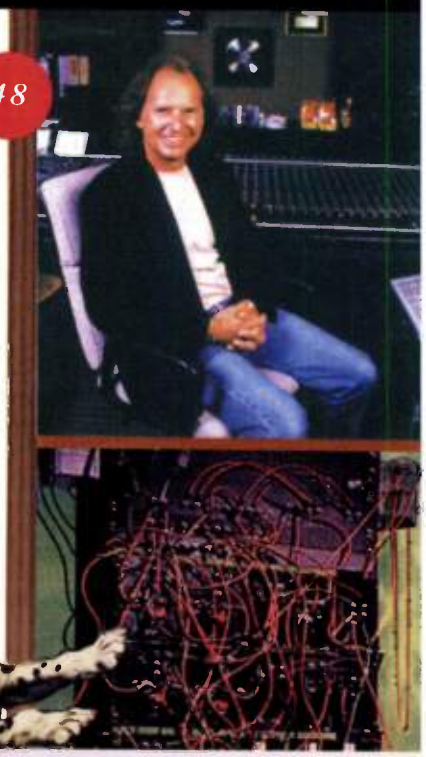
FEATURES

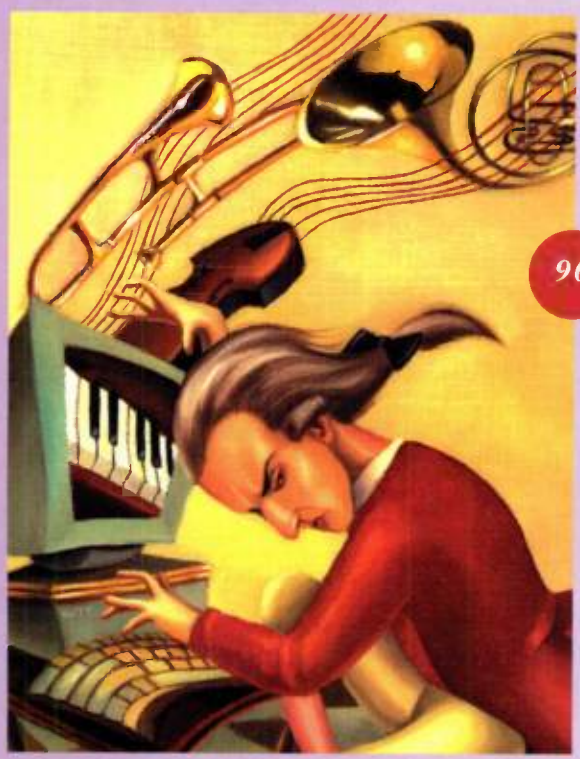


DEPARTMENTS

Electronic Musician®
OCTOBER 1995 VOL. 11, NO. 10

COLUMNS



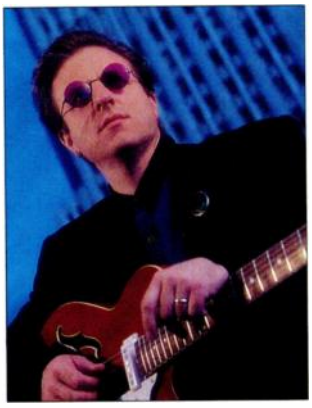


REVIEWS

Cover: Photo by Robert Perry.
Special thanks to BGW Systems, Inc., Carver Corp., Hafler Professional, and QSC Audio Products, Inc.

EM MIA?

In the online stampede, it's often better to step out of the way.



ROBERT PERRY

I've just about heard it all. The tantrums, queries, and opinions have flooded my e-mail box, buried our "Letters" editor, and crashed my voice mail. The discourse often spirals off into a zillion tangents, but the burning question is always the same: Why isn't **EM** online?

First, let me give you a lame answer. We *are* online and have been since way before the Internet's ascension to mass-culture hipness. It's not an elegant presence, but we've been "text talking" on PAN since 1987 (emeditorial@pan.com). In addition, there are **EM** folders on America Online in the Taxi Forum (keyword: taxi) and the Composers Coffeehouse (keyword: composers). And, of course, anyone who wants to bend my ear—or is it "eye"?—can send e-mail to emeditor@aol.com.

But, okay, I know what you're all *really* asking is why we don't have a Web page. Here's the simple, honest answer: We want to do it right. Now, please don't misinterpret this statement as flippant arrogance. Believe me, our online aspirations are driven more by intense self-doubt than the misguided confidence that whatever we do will be marvelous.

The bottom line is that it makes no sense for us to join the mad charge toward electronic publishing if we can't deliver a product that excites readers and enhances the print version of our magazine. We believe that an online publication is not a mere adjunct to a print mag, it is an entirely *different* product. You don't just upload reedited versions of the exact articles you've published in your print magazine. So, for us, this means we must develop an entirely new concept of information dissemination, one that preserves the style and personality of **EM**. It also means we must have the additional staffing in place to nurture this little jewel and maintain a consistent, interactive dialog with Net surfers. And it means that we must have the chops and commitment to provide content that best serves the needs of an online audience.

Adding to the dilemma (and delay) is the sad fact that there are few ideal publishing models we can study. Most of the online magazines I've encountered are real snoozers, and I'd rather go scuba diving with a school of piranha than establish a mediocre Web site. So, right now, we're all wearing our big overalls and pouring some concrete content into the foundation of *EM Online*. We're sweating over details and concepts and having a blast dreaming up exciting ways to deliver data. When we do take the online plunge, I'm certain you'll be happy with the results. Until then, thanks so much for your patience.

On the human front, I'm very happy to announce that we've hired Brian Knave as an assistant editor. Brian is co-owner of Moptop Records (a children's music label) and produces demos for bands and songwriters under the Knave of Arts Productions banner. He is also a professional drummer and drum teacher who has played with Norton Buffalo, as well as tons of local acts. Brian's rather spartan personal studio has already produced his label's first CD release, *The Search for the Moptop of Floppery Hair*, so I know he'll have a lot of great insights for all you dedicated home recordists. In fact, Brian is now in charge of our "Recording Musician" column. (Did I mention that he holds a Master of Arts in English from the University of California at Davis?) Someday soon you'll be seeing him, and the rest of the staff, on a delightfully informative **EM** Web page. I promise!

Michael Molenda

Electronic Musician®

Publisher Peter Hirschfeld

Editor Michael Molenda
Senior Editor Steve Oppenheimer
Technical Editor Scott Wilkinson
Associate Editors Mary Cosola, Larry Ullman
Assistant Editor Brian Knave
Copy Editor Diane Lowery
Editorial Assistant Jennifer Seidel
Contributing Editors Alan Gary Campbell, George Petersen

Art Director Linda Birch
Associate Art Director Linda Gough
Graphic Artist Dmitry Panich
Informational Graphics Chuck Dahmer

Associate Publisher Carrie Anderson
Southwestern Advertising Manager Dave Reik
Eastern Advertising Manager Angelo Biasi
Marketing Manager Elise Malmberg
Event Coordinator Jane Byer
Marketing Assistant Lena Inoue
Sales Administrator Joanne Zola
Advertising Sales Coordinator Christen Pocock
Sales Assistants Karen Dere, Tiffany Kendall, Ann Quirk
Classifieds Advertising Manager Robin Boyce
Classifieds Assistant Jef Linson
Classifieds Sales Assistant Jennifer Carrick

Director of Operations and Manufacturing Anne Letsch
Production Director Ellen Richman
Advertising Traffic Manager Shawn Yarnell
Production Assistants Michele Alaniz, Teri Stewart
Computer Systems Coordinator Daniel Lederer

Circulation Manager Steve Willard
Circulation Associate Karen Stackpole
Circulation Assistant Jeanette Campagna

Business Manager Benjamin Pittman
Receptionists Angel Alexander, Teresa Poss

Music and Entertainment Group
Group Publisher Hillel Resner

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

East Coast Advertising Office
tel. (203) 838-9100, fax (203) 838-2550

Subscription Services Office
(Address changes and customer-service inquiries)
PO Box 41525, Nashville, TN 37204
tel. (800) 843-4086 or (615) 377-3322

Cardinal Business Media, Inc.
1300 Virginia Dr., #400, Fort Washington, PA 19034

President and Chief Executive Officer
Robert N. Boucher
VP and Chief Financial Officer
Thomas C. Breslin
VP, Publishing Services R. Patricia Herron
VP, Human Resources Dorothy J. Flynn
VP, Sports and Music Division
Thomas J. Morgan
VP, Retail Division Marianne Howatson

Electronic Musician: (ISSN: 0884-4720) is published monthly by Cardinal Business Media, Inc., 6400 Hollis St., #12, Emeryville, CA 94608. One year (12 issues) subscription is $24; outside the U.S. is $49.95. Second Class 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.

Cardinal Business Media, Inc.
Also publishers of *Mix*® magazine.
Printed in the USA.

BPA INTERNATIONAL

now you can have a DP, too

DP/2 Features

2 24-Bit ESP Chips

2 Inputs/2 Outputs (balanced TRS)

65 Algorithms

600 Presets

DP/4+ Features

4 24-Bit ESP Chips

4 Inputs/4 Outputs (balanced TRS)

Process up to 4 Signals

54 Algorithms

400 Presets

have a DP/2!

With the resounding success of the DP/4+, you're probably wondering, "why doesn't ENSONIQ make a smaller DP for the home studio user *(like me)* on a budget?"

Well, enough of you asked, and we're happy to oblige – with the DP/2. A two-processor version of the DP/4+, for about half the price. *(Now you're talking!)*

The DP/2 can process one true stereo or two discrete mono sources, through two of the same custom DSP chips we use in it's big brother. It offers the same unequalled range and quality of algorithms as the DP/4+, plus new combinations designed specifically for the home/project studio. And we've built them into a whopping 600 Presets, ready for any challenge.

Both DP's feature balanced I/O, total programmability, advanced MIDI control, and even a headphone jack.

ENSONIQ's digital effects deserve the reputation they've gained – from studios, musicians, and the music press. For those who need the ultimate parallel effects processor, the DP/4+ is your best choice. But if you want that power in a smaller package, there's now a DP for you, too. *(Thanks!)*

You can reach us at:
800-553-5151 phone, 610-647-8908 fax
800-257-1439 documents to your fax
GO MIENSONIQ *on CompuServe*
http://www.ensoniq.com *on the Web*

ENSONIQ

LEADING THE WORLD IN SOUND INNOVATION

☐ Let me have more information on the DP/2!

Also send info on:

☐ DP/4+ *parallel effects processor*
☐ KT Series *64-voice weighted action synths*
☐ TS Series *synths* ☐ ASR Series *samplers*
☐ SQ Series *synths* ☐ KS-32 *weighted action synth*

☐ I'd also like to have your *make music* **think ENSONIQ** demo CD *(here's $5; ship it, handle it, whatever).*

Name _____

Address _____

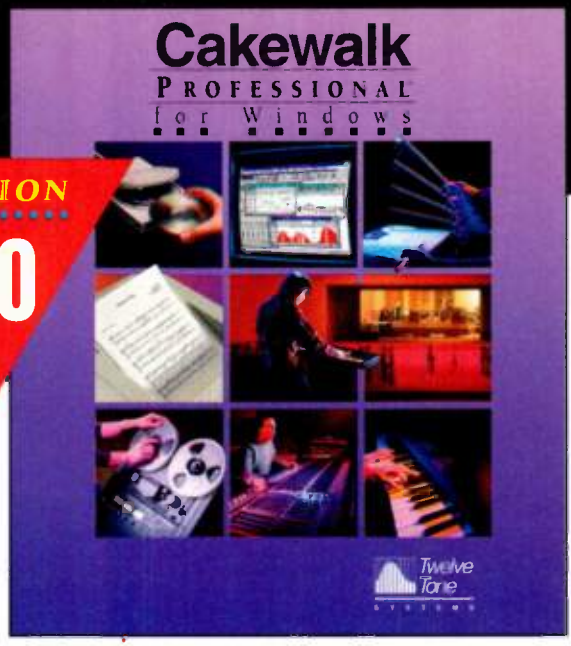
City _____ *State* _____ *Zip* _____

Phone (_____) _____

Mail to: ENSONIQ Corp, Dept. E-57, 155 Great Valley Parkway
P.O. Box 3035, Malvern, PA 19355-0735

think ENSONIQ

PLAY HARD.



WORK EASY.

Tired of music software that's hard to work with? Spending more time making tech support calls than you are making music?

Then bring in the Professional.

Cakewalk Professional remains the leading MIDI sequencer for Windows today. It's powerful, fast, stable, and yes — extremely usable.

And while other music software companies scramble to release something on the Windows platform, Twelve Tone Systems is now shipping the third major release of its award-winning sequencer.

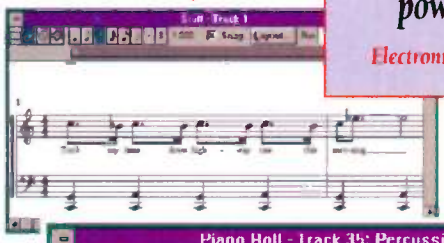
Here are some of the new 3.0 features:

BANG ON THIS

So what *else* is new in 3.0? Plenty.

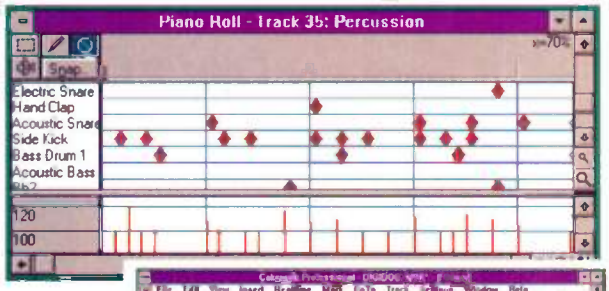
- Percussion Editing
- MIDI Machine Control
- Enhanced Swing Quantizing
- Printing Up to 24 Staves per Page
- Expanded Instrument Definitions
- Bank Select
- Way More

> *"Simply put, it's a professional powerhouse."*
>
> Electronic Musician, 3/94



GRAB A GROOVE

The new Groove Quantize option lets you "steal the feel" of one track, and use it to quantize another. Cakewalk's own groove format supports note start-times, durations and velocities. Also works with DNA Grooves.



WHAT'S THE WORD

Add lyrics directly into the Staff view, and print in your notation. For on-stage performances, use the Lyrics view to see scrolling lyrics or stage cues in large fonts.

MASTER MIXES

Mix volume, pan and other controllers using 96 assignable faders and 32 Note On buttons. Create fader groups for automated cross-fades and mix-downs. And the Faders view now fully supports the Mackie OTTO 1604 MIDI automation package.

ELECTRONIC MUSICIAN 1995 EDITORS' CHOICE

GENERAL MIDI

PC Multimedia PC

OTHER PROFESSIONAL FEATURES:

256 tracks; rock-solid SMPTE/MTC sync; custom programming language with macro recorder; MIDI remote control; system exclusive librarian and event filters, to name a few.

And with support for MCI commands and digital audio wave files, Cakewalk Professional can be the engine that drives your next multimedia project.

So play all day. Play all night. And let Cakewalk Professional do the hard work for you.

NOW PLAYING EVERYWHERE

Cakewalk Professional 3.0 is just $349, and is available at finer music and computer stores everywhere. For more information, or to order the Cakewalk Professional Demo Pack for just $5, call:

800-234-1171 OR 617-926-2480.

Call today and get a free copy of Twelve Tone Systems' customer newsletter, *QuarterTone*, while supplies last.

Twelve Tone SYSTEMS

P.O. Box 760, Watertown, MA 02272

EVS Effective Video Solutions

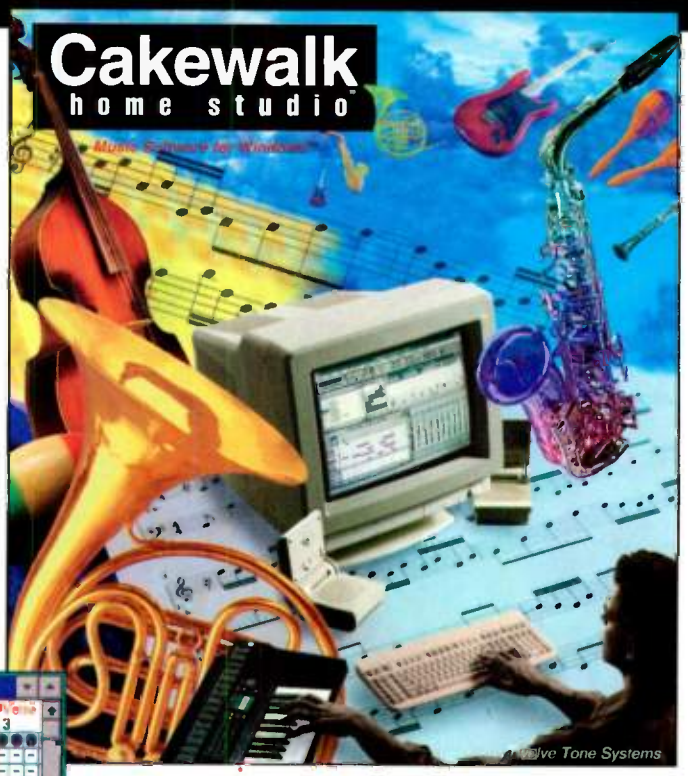
Video Tutorials Available
Call 1-800-234-1171

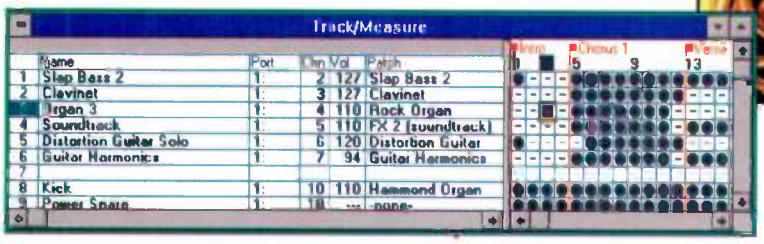
Turn On. Play In. Print Out.

It's that simple. If you want a great MIDI sequencer for Windows, but you don't need all those pro-level features, then you should get into Cakewalk Home Studio™.

Cakewalk Home Studio is a great way to start exploring the world of music and computers.

Once you turn it on, you'll find many of the essential recording and editing tools that are part of the award-winning Cakewalk Professional™ software.





Get Started

Cakewalk Home Studio has everything you need to start creating music with your PC: 256 tracks; Staff, Piano Roll, and Event List views; a 16-track Faders view; and high-quality notation printing.

The Home Studio edition also plays back digital audio wave files on Windows-compatible sound cards. So you can add sound effects or voice directly into your MIDI sequence.

Take the Next Step

And when you think you're ready, expand your MIDI experience by upgrading to Cakewalk Professional. Although it's a more advanced sequencer, Cakewalk Professional includes all the views and commands found in Home Studio. So making the move to the Professional edition will feel natural.

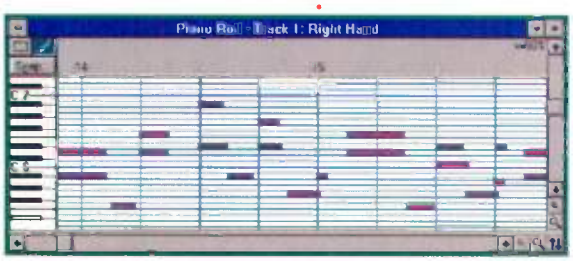
Cakewalk Home Studio Highlights:

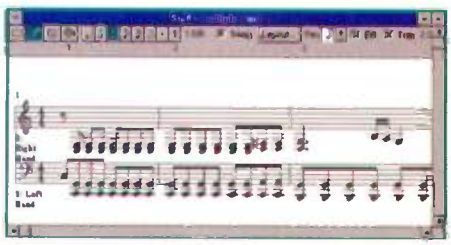
- 256 tracks
- Real-time and step recording
- Track/Measure, Piano Roll, Event List, Staff and Controller views
- Faders for volume, pan, and reverb
- Drag-and-drop editing
- Event Filters for criteria edits
- Graphical tempo map
- Meter/key map
- Markers for organizing song parts
- Plays wave files on Windows sound cards
- Displays General MIDI, other instrument patch names
- Prints up to 24 staves per page with lyrics

Now Only $129!

1995 Multimedia World Readers' Choice Winner

PC Multimedia PC





Try It Today

So drop into your local music or computer store, and tell them you want to get started with Cakewalk Home Studio. For the name of an authorized Cakewalk dealer near you, call Twelve Tone Systems at

800-234-1171 or 617-926-2480.

Demo pack for just $5. Call today and get a free copy of the Twelve Tone Systems customer newsletter, *QuarterTone*, while supplies last.

Twelve Tone SYSTEMS

P.O. Box 760, Watertown, MA 02272

EVS Effective Video Solutions

Video Tutorials Available
Call 1-800-234-1171

LETTERS

MR. FIX-IT DOES IT AGAIN

I've been a subscriber since 1987. (I have two file boxes with back issues that I lug around whenever I move to prove it.) Thanks for the "Service Clinic" tip regarding the cleaning of a Multimoog ("Tricks of the Trade," August 1995). I just moved to Raleigh, North Carolina, and I hadn't played mine for some time. When I fired it up, it sounded terrible. Remembering that I had read about cleaning Multimoogs in **EM**, I turned it over, popped the covers off, cleaned it, and tried it again. No noise, just that sweet, fat Moog sound we all know and love.

Frank Tornoe
Raleigh, NC

WARP SPEED

In the August 1995 issue, Charles Brannon gave us a preview of Windows 95 and its possibilities for musicians ("Windows 95 Preview"). As always, the writing in **EM** is excellent and insightful regarding new products. However, is it fair that only one operating system got reviewed? Microsoft has promised its new 32-bit operating system for quite some time now—remember Chicago? It took so long for Microsoft to develop it they had to change its name to Windows 95.

If **EM** wants to review a new operating system that can support Windows 3.1 as well as DOS applications, one that is a true multitasking/multithreaded platform and is crash protected, one that can fully exploit the 32-bit capabilities of the 80386 (and higher processors), then why isn't IBM's OS/2 Warp reviewed? After all, OS/2 Warp has been shipping since December 1994; Windows 95 is probably looking at another delay.

Paul L. Dial
Houston, TX

Paul—Although we cover computer products and issues that affect musicians, we are not a mainstream computer magazine like PC World, PC Computing, *or* Byte. *When we write about computer operating systems, we focus specifically on how they affect music software and music production. OS/2 Warp is not a significant player in this market; the vast majority of our PC-based readers use Windows 3.1 (though a few still use straight DOS), and I think most will quickly move to Windows 95.*

Many music applications are being updated to take full advantage of Windows 95's advanced features, so we felt it deserved our editorial attention. In contrast, despite OS/2 Warp's advantages over Windows (and even Windows 95), virtually no music applications have been written for it, and there is no reason to believe this will change in the foreseeable future.

Sure, you can run Windows applications within OS/2 Warp, but then you're just running Windows on top of another operating system. You'll get better crash protection, and you'll be able to use preemptive multitasking, but the music programs won't run any better. Because OS/2 Warp's specific advantages for musicians are few, we have not devoted editorial space to it.—Steve O.

IS THAT SO?

As a conservative Republican, I am getting fed up with people taking my money and using it to criticize my political party. The government takes my taxes and gives it to National Public Radio so they can bash conservatives. My union buys anti-Republican ads in the *Washington Post*. And now the magazine I relied on to bring me the latest news on MIDI and electronic instruments finds it necessary to join the crowd ("The Front Page: Enhanced Boredom," August 1995). To say that Senator Robert Dole has not seen the things he objects to is to deny reality. For example, take violence on TV. I do not own a TV, but I see numerous acts of violence on TV as I pass the displays in department store windows or sit in the waiting rooms of doctors' offices. Please cancel my subscription.

Osborne C. Tasker
Harpers Ferry, WV

NET WORTH

Thanks for the article on finding the hits online ("Online Juke Joints," July 1995). I am relatively new to the Net, and both this article and the one in the October 1994 issue ("Magical Musical Tours") have been a great help. Keeping up with technology has been a lot easier since I subscribed to **EM**.

Todd Masten
Boston, MA

THE TRUTH ABOUT ALEX

I was excited to see your article about the tribute album to The Sensational Alex Harvey Band (July 1995). I am a big-time Alex Harvey fan, and I wanted to share some of my experience with you. When I was a deejay at my college radio station, I'd get a stack of albums every day and listen to more music than I ever dreamt possible. One of the albums I checked out was The Sensational Alex Harvey Band. It was their first album, called *Next*. I *hated* that album. I played it over and over in amazement that anyone would have the nerve to record it and that a record company would put it out. I played it every day. I played it for friends. I even played it on the radio.

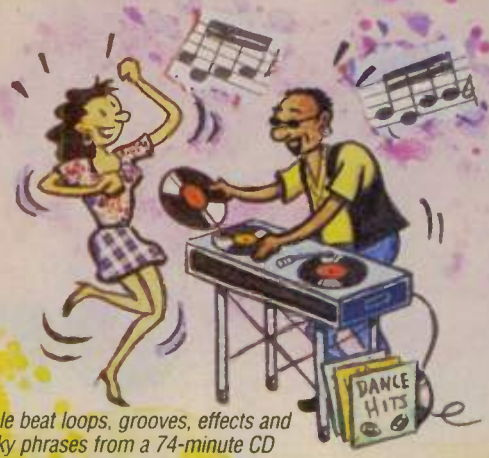
Then one day, someone said to me, "You really like this band, don't you?" Red-faced and shocked, I denied it, claiming that Harvey had destroyed the word "sensational." Total denial! When I played *Next* that evening, I realized I was a fan. Big time.

VALERIE SPAIN

INSTRUCTIONS:

Buzz just bought a new Roland MS-1 Digital Sampler. Now he's dying to see how versatile his cool new purchase is. See how many ways you can get Buzz to work his MS-1 before he kicks back at home with his pal, Spot.

START



Sample beat loops, grooves, effects and funky phrases from a 74-minute CD that comes with every MS-1.

Play mondo drum sounds or any sound from a MIDI percussion controller.

Standard, non-volatile memory provides 49.2 second sampling time so you can play along with licks and phrases. With an optional PCMCIA card, you can go beyond 20 minutes.

Eight easy-to-use pads with multiple banks trigger sound effects instantly.

With stereo sampling based on 16 bit A/D/A conversion and onboard sequencer, the MS-1 can cut it in the studio.

FINISH

If you got Buzz to all his stops, you probably picked up on something along the way. For beat loops, remixes, studio work, drum sounds, sampling and live performances nothing gets you going like the MS-1. With unlimited versatility and a tiny price tag, you'll never find yourself at a creative dead end. Move on down to your nearest Roland dealer and try one out. It'll amaze you.

See us at AES Booth #1105

Roland

Roland Corporation U.S., 7200 Dominion Circle, Los Angeles, CA 90040 (213) 685-5141
Roland Canada Music Ltd., 5480 Parkwood Way, Richmond, B.C. V6V 2M4 (604)270-6626 COMPUSERVE GO ROLAND

WRH

Korg changes the rules:

One day, every music workstation is going to be like this.

TRINITY
MUSIC WORKSTATION DRS

Presenting Korg Trinity. Years ago, our first music workstation revolutionized what keyboards could do. Now we'd like to revolutionize the revolution.

Trinity™ is a series of new keyboards, re-thought from top to bottom. Each has a groundbreaking modular design, so even if you start with the basics (two fully loaded sound banks and a powerful 16-track, 60,000-note sequencer), you can turn it into a total digital recording environment – with a mind-blowing range of sounds and effects.

The 61-note Trinity and Trinity Plus both feature a new tone generation system that plays back at 48kHz, with a huge 24MB of PCM ROM. A Flash ROM option lets you load in sounds from Korg and Akai libraries. And far better than mere sound boards, Trinity lets you add an entirely new synthesizer: The Prophecy expansion board (standard in Trinity Plus) gives you DSP models of synths and acoustic instruments.

You also have an option for 4-channel hard disk recording, letting you record any sound – using Trinity's built-in effects processors if you wish.

The only thing more amazing is that you can control it all with one finger. Trinity features a large touch-sensitive screen (and a ribbon controller, too).

In your lifetime, you'll probably only see a few keyboards that literally change the way you work and play. So savor this one. Korg Trinity is here.

KORG® TRINITY
MUSIC WORKSTATION DRS

- *24MB internal PCM ROM (Bank A, Bank B).*
- *Multi-mode digital resonant filters.*
- *114 dynamic multi-effects; 8 timbre insert effects and 2 master effects available simultaneously.*
- *Built-in 16-track, 60,000-note sequencer.*
- *Large TouchView™ Graphical Interface.*
- *Dynamic Ribbon Controller, joystick, assignable switches, pedal and foot controllers.*
- *Optional 8MB PCM Flash ROM for sounds from Korg and Akai libraries (Bank C, Bank D).*
- *Optional Prophecy Expansion Board (Bank S) adds second synthesizer based on DSP sound modeling synthesis. Contains 64 programs (128 with Flash ROM option).*
- *Optional 4-channel hard disk recording system.*
- *Four polyphonic outputs, plus optional SCSI, SPDIF and Alesis I/O ports.*
- *76-note Trinity Pro and 88-note, weighted-action Trinity ProX coming soon!*

A year later, I got to see them in concert. The audience booed, yelled, and threw things at SAHB. Harvey, with his great sense of humor, brought everything down low behind him, asked for the house lights to be turned on, and said, "I'm getting what I would call a mixed response from this audience. If you're not enjoying what we're doing up here, I want you to get up on your chair and do this." He held his hands up and flashed the finger with each hand. The place went up for grabs; the crowd was up on their chairs giving them the finger. The band took that energy from the audience, went into warp drive, and never flinched from putting on a great show.

Thanks for bringing back some great memories. In all my years of being a fan, I've only met one other person who knew them, so this was a special treat for me. I wish there were CD releases of the SAHB albums. Maybe your tribute album will make that happen. I know I'll buy one, and I'll make a point of bugging all my friends with it.

Victor Przysiezny
Highland Park, IL

SWEETENING POP

Thanks for an informative and entertaining article on Richard Carpenter in the May 1995 issue ("Pop Charts"). Arranging is an area many magazines overlook in favor of articles on recording techniques, which, while useful, are in ample supply. Although my band has currently confined itself to 8-track demos, we spend a lot of time working on adding "the little things" that subtly enhance a recording.

I've always been amazed at how much sweetening you can do (and the variety of flavors you can get) with a couple of shakers and a tambourine. In the year since we started seriously recording, I've noticed how those two instruments seem to show up everywhere in commercial music. I've discovered that a great guitar solo sounds even better with an egg shaker driving it along and that a song's chorus brightens up just a bit more with a tambourine behind it. I've also found that hip hop and rap records are great sources of ideas for percussion enhancements, because the basic drum beats often vary little over the course of the song and changes are provided by additional instruments coming in or dropping out. Obviously, percussion is only the tip of an extremely varied instrumental iceberg.

Michael Coffman
coff5787@mlb.com

BUZZ BUZZ BUZZ

When my VGA monitor is on, I hear a loud buzzing sound in my studio speakers and headphones. I have been getting around this problem by turning off my monitor when I record. Can you tell me how to fix the buzzing?

ibot774@aol.com

Ibot774—Unfortunately, you haven't provided much information for me to go on. There are several possibilities. The first thing I would do is turn off all inputs to the mixer and see whether the buzz disappears. If so, turn on the inputs one at a time and listen for the buzz; this will identify which inputs are affected. You might have routed some audio cables near the monitor, its power cord, and/or its video cable. It could also be a grounding problem; for more info

on grounding issues, see "On Solid Ground" in the September and October 1992 **EM**. *Make sure the video monitor is not located close to the speakers or other audio gear, especially if your speakers are not shielded. I have also heard of video monitors inducing noise into nearby microphones or guitar pickups; if you have either of these devices near the monitor, try moving them as far away as possible.—Scott W.*

WORLDLY FEEDBACK

I've wanted to write to you since last year, but I was thinking, "Why bother? They know they're good; my letter won't bring them anything new." Then I realized you may want some feedback from this part of the world.

EM is great. It features good articles, covers a wide variety of music-related topics, and keeps us updated with the latest in software and equipment. It is pleasant and relaxing to read and, at the same time, is genuinely informative. In my opinion, however, **EM**'s main strength is the high quality of its writing. Other specialized periodicals or books on electronic music may be fine technically, but the quality of the writers and contributors to **EM** is what makes it really stand out.

A computer specialist by trade, I'm also a pianist, guitarist, composer, singer, and home recordist. In my personal studio, I make extensive use of MIDI gear and synths. I've had a regular subscription to your magazine for many years. On several occasions **EM** has helped me do better recordings, write better music, and put me in contact with equipment suppliers in the United States. We do not have many professional music outlets here in Jordan and the addresses I find in **EM** often come in handy.

You probably think I'll end this letter with a comment or remark of some sort, telling you what **EM**'s weaknesses are, what I recommend you do to correct them, and so on. Well, I won't. Whatever weaknesses there might be, they must really be minor ones—after all, nobody's perfect. I just want to say thanks **EM**, reading you is such a pleasure. Give me more of the same.

Jean-Claude Elias
Amman, Jordan

ERROR LOG

August 1995, "Vestax HDR-6," p. 121: In the "Gripes" section, the reference to "having other unrelated tracks pop up during a take" should have been deleted, as this "problem" proved to be the result of user error.

August 1995, "Gregory Paul Productions George Clinton Sample Series," p. 128: The correct telephone and fax numbers for AEM Record Group are tel. (810) 977-2053; fax (810) 977-0357.

September 1995, "What's New," p. 16: Temporal Acuity Products has a new phone number and fax: tel. (800) 99-PIANO or (206) 881-9797; fax (206) 881-9664.

WE WELCOME YOUR FEEDBACK.

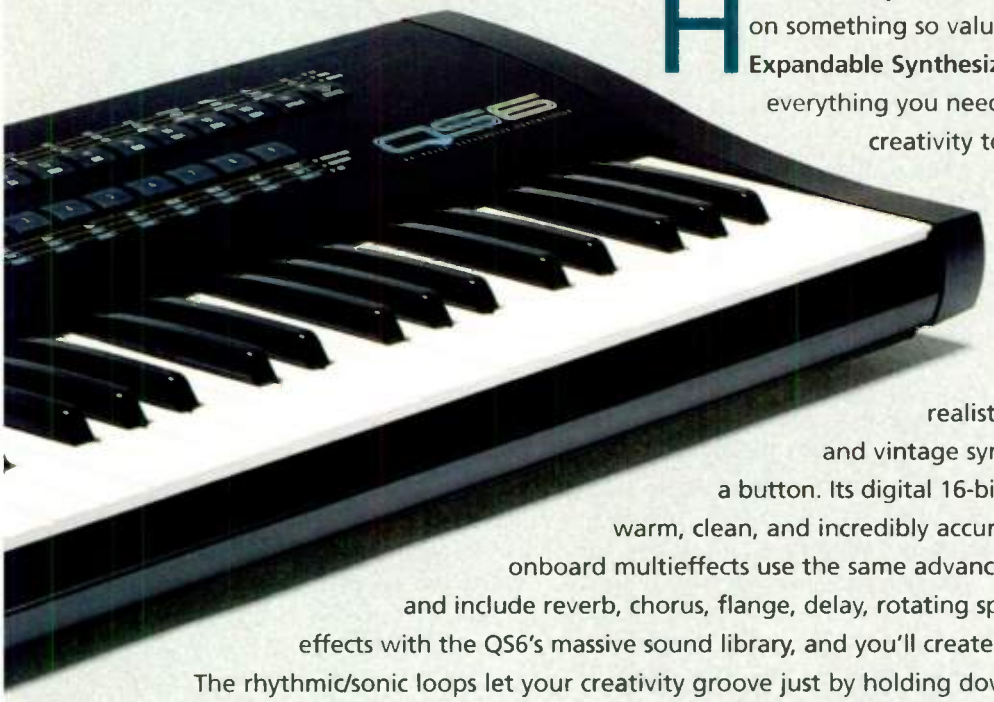
Address correspondence to "Letters," Electronic Musician, 6400 Hollis St., Suite 12, Emeryville, CA 94608 or e-mail at emeditorial@pan.com. Published letters may be edited for space and clarity.

Creativity Is Priceless. Now, It's Also Affordable.

Introducing The QS6™ From Alesis

QS6

64 VOICE EXPANDABLE SYNTHESIZER



How much is your creativity worth? You really can't put a price on something so valuable. The powerful new **QS6™ 64 Voice Expandable Synthesizer** gives you everything you need to push your creativity to new levels. It's the affordable solution for musical inspiration.

Studio Quality Sound

The QS6 has all the great sounds that inspire you to do your best work. You'll find everything from realistic acoustic instruments to cutting-edge and vintage synth sounds, all accessible at the touch of a button. Its digital 16-bit linear non-compressed samples provide warm, clean, and incredibly accurate voicing for every performance. The onboard multieffects use the same advanced processor as our Q2™ Master Effects, and include reverb, chorus, flange, delay, rotating speaker effects and more. Combine these effects with the QS6's massive sound library, and you'll create CD-quality mixes right from your synth.

The rhythmic/sonic loops let your creativity groove just by holding down a single key. QS6's great feeling, semi-weighted keyboard offers velocity, release velocity and aftertouch for expressive playability. Furthermore, all of this power is packed into a tough all-metal chassis that will take the punishment when you take your show on the road.

Choice, Variety and Expandability

The QS6 offers a huge palette of internal sounds - 8 megabytes of sample waveforms, 640 Programs, and 500 Mixes - giving you thousands of timbral options to choose from. This sonic library can be expanded instantly by simply plugging in a 4MB or 8MB Alesis QuadraCard™, like our acclaimed Stereo Grand Piano card. Also, the QS6 includes Alesis' exclusive Sound Bridge™ software (for Mac and PC) which allows you create custom cards with your own sound files from your computer. This innovative technology guarantees that you'll always have access to new sounds.

Powerful Synthesis Engine

The QS6 uses the same advanced synthesis architecture as its big brother, the QuadraSynth Plus™. True 64-voice polyphony lets you assemble complex sequences and rich, stacked chords. Its 16 channel multitimbral Mixes and a built-in computer interface (also for both Mac and PC) give you easy access to the world of MIDI sequencing, software and composition. In fact, we've included a free CD-ROM with the QS6 that's packed with extra sounds, killer sequences, and Steinberg's Cubase Lite™ sequencing software to help get your creative juices flowing.

The QS6 is an instrument that was crafted to help you unlock your creative talents. Stop by an Alesis Dealer and start creating new music with the newest keyboard from Alesis.

For more information about the QS6, see your Authorized Alesis Dealer or call 310-841-2272
QS6, QuadraSynth, QuadraCard and Sound Bridge are trademarks of Alesis Corporation

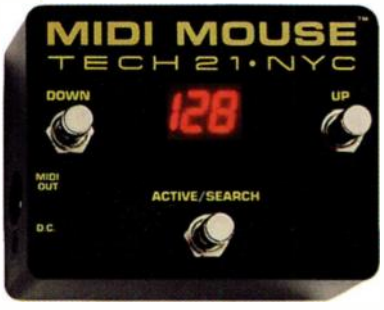
Alesis Corporation 3630 Holdrege Avenue Los Angeles CA 90016 310-841-2272 alecorp@alesis1.usa.com

ALESIS

See us at AES Booth #556

WHAT'S NEW

By Steve Oppenheimer



▲ TECH 21 MIDI MOUSE

When it's time for a (program) change, Tech 21 would like you to put your foot in it! To be specific, the company wants you to stomp on the MIDI Mouse ($125), their new MIDI Program Change controller. The handy little 3.5 × 4.5-inch stomp box has just three footswitches and a large, nonglare, 3-digit LED display. Ease of use should not be an issue here.

The Active/Search switch toggles between the unit's two operating modes. In Active mode, the Up and Down switches step sequentially through the 128 Program Changes, which are sent on a selectable MIDI channel. In Search mode, the Up and Down switches scroll through the program numbers, but the Program Change command is not sent until you switch back to Active mode.

The LED display flashes in Search mode and remains steadily lit in Active mode. The MIDI Mouse can run on a 9V battery or an AC adapter. Tech 21; tel. (212) 315-1116; fax (212) 315-0825.

Circle #401 on Reader Service Card

▶ E-MU DARWIN

Drawing on its extensive experience as a leading designer of digital samplers, E-mu Systems has introduced DARWIN, an 8-track digital-audio hard-disk recorder ($3,195). The unit supports MIDI Machine Control and MIDI Time Code, records at 44.1 or 48 kHz with 16-bit resolution, and can operate at -10 dBV or +4 dBm levels. The unit is available with an optional 1 GB internal hard drive ($3,795), and a 50-pin SCSI port is provided for external data storage.

The stand-alone device is intended to provide easy-to-use, nondestructive editing. The user interface includes tape recorder-style transport buttons, 10-segment LED level meters with clip and peak-hold indicators, and a jog/shuttle/data wheel. The 240 × 64, graphic LCD screen displays SMPTE times, and it features named tracks with audio regions, which are displayed as bars that can be viewed at various Zoom levels. Editing features include cut, copy, paste, and move, with multiple levels of undo and redo. All editing and recording features are accessed from one Edit screen.

An Audition Edit mode plays back your nondestructive edits, and a Rehearse function lets you practice destructive punches before you commit. Punches can be accomplished tape-recorder style with a footswitch. In addition, 40 autolocate points help you get back to the desired spots quickly.

A Virtual Slave Reel (VSR) feature lets you create different versions of a project without altering the original data. VSR also lets you nondestructively bounce and recombine tracks internally to create new tracks and submixes. The onboard 8 × 2 digital mixer includes panning and lets you create a separate headphone mix during internal bounces.

DARWIN has eight balanced (TRS) ¼-inch analog outputs and stereo, S/PDIF digital I/O. It comes with four balanced ¼-inch analog inputs, and an optional 4-input expansion card lets you record up to eight tracks simultaneously. The unit's five rear-panel ports let you add the input expander, an ADAT digital I/O interface, a SCSI host port card for connection to an external computer, and a sync card for linking up to four DARWINs. Prices and ship dates for the expansion cards were not announced at press time. E-mu Systems; tel. (408) 438-1921; fax (408) 438-8612.

Circle #402 on Reader Service Card

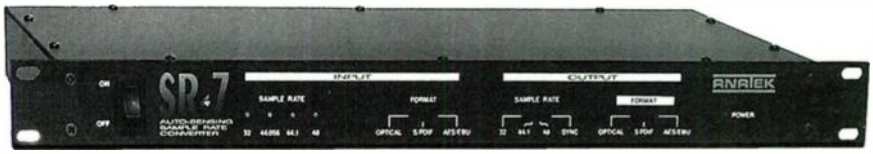
▼ ANATEK SR-7

Anatek is best known for its Pocket Product series of MIDI accessories, but the Creation Technologies subsidiary has expanded its line in new directions. Anatek's latest product, the SR-7 ($599), converts between AES/EBU and S/PDIF audio formats and between 32, 44.056, 44.1, and 48 kHz sample rates.

XLR, RCA, and optical I/O connectors (selected with a front-panel switch) are provided for compatibility with most digital-audio devices. A pair of switches set the desired output format and sample rate. (The sample-rate switch also can select external sync to AES/EBU or DARS.) A set of LEDs show the sample rate of incoming audio.

The SR-7 operates with 20-bit resolution and uses a new algorithm that is said to virtually eliminate conversion distortion. In addition, input signals that exhibit jitter are stabilized, and SCMS copy-protection can be blocked. The manufacturer claims a THD+Noise spec of -108 dB and a dynamic range of 120 dB. Anatek; tel. (604) 430-4336; fax (604) 430-4337.

Circle #403 on Reader Service Card



ALESIS QS6

Alesis introduced several items of interest at this year's Summer NAMM show. Heading the slate is the QS6 synthesizer ($1,099), a compact, QuadraSynth-series keyboard instrument in a roadworthy, all-metal chassis. The QS6's 61-key, semiweighted keyboard supports Velocity and Release Velocity as well as Channel Pressure. A rear-panel serial port can be switched between RS-232 and RS-422 for connection to a PC or Mac. The synth has pitch and modulation wheels, a control slider, and inputs for a sustain pedal and an assignable CV/switch pedal.

Like the original QuadraSynth, the QS6 is 64-voice polyphonic and 16-part multitimbral. It has 8 MB of waveform ROM, as well as 512 Programs (including a full bank of General MIDI sounds) and 400 multitimbral Mixes in ROM. In addition, there is enough RAM for 128 user Programs and 100 user Mixes. A PCMCIA expansion slot accommodates 8 MB QuadraSound sound cards.

Alesis bundles a mixed-mode CD-ROM containing various third-party software (notably Steinberg's *Cubase Lite* sequencer for Macintosh and Windows), General MIDI sequences, and extra Programs. Also on the disc is Alesis' *Sound Bridge* software, which lets you import your own samples and Audio Interchange File Format (AIFF) sounds into the QS6. The imported samples can be saved to optional Flash RAM cards.

Hoping to build on its success with drum machines and the D4 drum module, Alesis also introduced the DM5 drum module ($449), which features 18-bit DACs. The 1U rack-mount module includes over 500 drum and percussion sounds, including new kicks, cymbals, snares, toms, hi-hats, and assorted percussion and special effects. Some of Alesis' greatest "hits" from previous machines are also included.

The DM5 has four audio outputs and twelve trigger inputs. There are five programmable parameters for each trigger, and activity at each input is indicated on the unit's backlit LCD. The unit supports MIDI Volume (CC7) and offers user-programmable panning (seven positions) and Dynamic Articulation, which lets you modulate the timbre and pitch with MIDI Velocity.

Alesis also incorporated a few features that are new to its product line. The Random Sample feature randomly switches between similar samples (say, hi-hats) for a more realistic performance. When you exceed the unit's 16-voice polyphony, the MIDI Overflow feature sends the excess notes to the MIDI Out port for routing to another sound source.

With Alesis' MidiVerb 4 and QuadraVerb 2 already in the field, it should be no surprise that the MicroVerb 4 ($299) is here, too. The new unit includes 18-bit Delta-Sigma converters and offers a frequency response of 40 Hz to 20 kHz with a dynamic range of >90 dB.

The MicroVerb 4 includes multi-effects algorithms that employ up to three effects—a first for MicroVerbs—many of which are based on the algorithms in the MidiVerb 4 and QuadraVerb 2. The 1U rack-mount device offers a total of 100 presets and 100 user programs, including assorted reverbs, delays, chorusing, flanging, and rotating speaker effects. Many effects are in discrete stereo.

Alesis has tried to retain the traditional MicroVerb ease of use. For example, each program has two parameters that can be edited in real time using two front-panel knobs. The unit responds to MIDI Modulation messages and Program Changes, which are displayed on a numeric LED display.

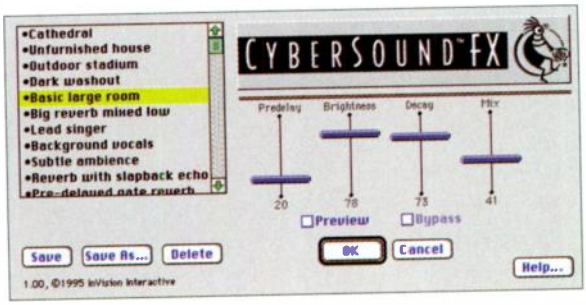
A ¼-inch TRS footswitch jack accepts a dual footswitch that lets you step through programs and switch in and out of Bypass mode. Alesis; tel. (310) 558-4530; fax (310) 836-9192; e-mail alecorp@alesis1.usa.com.

Circle #404 on Reader Service Card

INVISION CYBERSOUND FX

It seems DSP plug-in software that adds functionality to a central program is all the rage these days. Many computer-based musicians know about the many plug-ins for Digidesign software and hardware. But multimedia musicians should also know that Adobe's *Premiere* has a plug-in architecture, which has also been adopted in OSC's *Deck II* 2.5 hard-disk recording software. The popularity of these programs makes the release of third-party *Premiere* plug-ins a significant development.

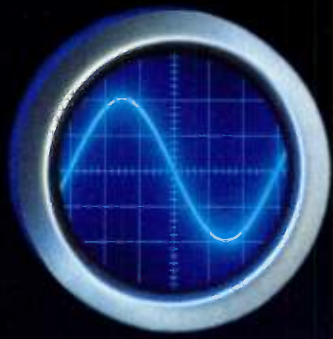
CyberSound FX ($129) is InVision Interactive's first move into this burgeoning market. The audio DSP plug-in produces seventeen effects, including reverbs (chamber, small hall, and large hall), delays (echo, single, and multitap), chorus, phaser, flanger, pitch shift, parametric EQ, shelving EQ, dynamic filter, tremolo, wah, compression, and gain normalization.



There are more than 200 factory effects templates designed for a variety of common applications. The effects are also fully programmable, and user settings can be saved with names and comments.

The program requires a 68020 or better Macintosh, 4 MB of RAM, QuickTime 1.6 or later, and Adobe *Premiere* 3.0 or later. *CyberSound FX* runs in native code on a Power Macintosh. InVision Interactive; tel. (800) 468-5530 or (415) 812-7380; fax (415) 812-7386.

Circle #405 on Reader Service Card



AUDIO INTEGRITY

You are busy trying to create the perfect mix, so we'll get right to the point. The new Sony DPS-V77 is a single rack powerhouse – a digital multi-effects signal processor that combines the best of our DPS Series, for an impressive array of effects including reverb, delay, modulation, dynamic filtering and more. The V77 also offers balanced and unbalanced analog and digital I/Os, with high resolution 24-bit A to D, 20-bit D to A converters and Sony's proprietary 32-bit digital signal processing. Result: great sound in, better sound out.

DUAL EFFECTS ARCHITECTURE

With 50 effects per block, the dual block architecture of the DPS-V77 makes it extremely flexible, since it allows for various serial and parallel configurations. Each block is equipped with a switchable pre or post effect EQ. You'll find 198 user presets in addition to 198 factory presets. You'll also discover several new, ear-opening effects, from intelligent pitch shifting to irregular delays, to mono/stereo conversion, and three-dimensional spatial placement. The most important feature, however, may be what this unit *doesn't* come with.

IT'S A SIGNA

IT'S A GARBA

MORPHING

Nietzsche once said, 'get rid of the junk you don't need.' Or something like that. Anyway, sound philosophy for life is sound philosophy for signal processors. So, that annoying drop-out you get switching

L PROCESSOR.

GE DISPOSAL.

between effects? It's history. Eliminated. Toast. Seamless transition – say, between the tail end of a reverb into a chorus – is now reality. With this 'morphing' function of the DPS-V77, a new effect (like a chorus or flange) can begin while a current effect (like a reverb or delay) is decaying, giving you from 0 to 10 seconds transition time.

USER FRIENDLY

Take a good look: nice, big LCD display and "jog/shuttle" knob. Numeric key pad, descriptive icons. Assignable direct access keys, located where they ought to be. Call up a preset and up to 6 of your most frequently used parameters are there instantly. All this, plus full MIDI control. So work flows quickly, smoothly, easily.

As easy as calling 1-800-635-SONY, ext. DPS, for more information.

Imagine. No more fumbling, grumbling, mumbling with a whole rack of trouble. No more extraneous junk. If only the rest of your problems were this easy to get rid of. You know, run the water, switch, and whoosh. Down the drain in one last gurgle. Goodbye bills, goodbye mortgage, goodbye mother-in-law, goodbye back pain.

SONY

▶ BLUE CHIP AXON NGC-66

Pitch-to-MIDI converters are big request items with **EM** readers. Also high on the wish list is a better-tracking MIDI guitar, especially if it does not require a computer. But previous attempts at hardware pitch-to-MIDI conversion have generally fallen short of the mark, and most MIDI guitars still suffer from tracking-speed limitations.

Blue Chip Music Technology's AXON NGC-66 Neural Guitar MIDI Controller ($1,195.95) uses DSP-based, neural-net technology to analyze the pitch of a plucked string. Among the many advantages of this approach is the ability to detect where the string is fretted. This lets you do keyboard-like splits at any fret and assign different MIDI channels to the zones on either side of the fret. The AXON also senses your pick position and lets you assign up to three different zones based on the pick's location between the bridge and neck.

According to exclusive distributor Music Industries Corp., the AXON can usually analyze a plucked string in 5 to 8 milliseconds and never takes more than 14 ms. This compares well with competing units, which may require 24 to 40 ms for detection, causing a noticeable tracking lag. Because the unit does not have to wait for the string to complete one cycle—most MIDI-guitar converters must analyze two cycles—it works with bass guitar. The AXON uses a Roland GK-2A or compatible pickup (not included). Music Industries Corp. (distributor); tel. (800) 431-6699 or (516) 352-4110; fax (516) 352-0754.

Circle #406 on Reader Service Card

▼ TASCAM DA-P1

Alone in the field, where no one can hear you scream, your portable DAT recorder had better be sturdy and reliable. According to TASCAM, their DA-P1 ($1,899) is all that and more, with a rugged transport that features two direct-drive motors. The audio hardware includes two heads and new-generation, Delta-Sigma A/D and D/A converters. Three sample rates—32, 44.1, and 48 kHz—are supported, and digital copies can be made without SCMS copy-protection.

The unit has XLR mic/line inputs with phantom power, along with line-level analog I/O on RCA connectors. S/PDIF digital I/O is also provided on RCA jacks. A mic limiter and 20 dB pad help control record levels, and a headphone jack with level control is included for convenient monitoring.

The DA-P1's display shows program time, sampling rate, ABS time, and record margin. According to the manufacturer, functions such as counter mode, counter reset, margin reset, and ID select are easily accessed. The machine comes with a shoulder belt, AC adapter, and one battery. Options include additional rechargeable batteries and a carrying case. TASCAM; tel. (213) 726-0303; fax (213) 727-7656.

Circle #407 on Reader Service Card

▶ SONIC FOUNDRY SOUND FORGE XP

Having established its credentials with the powerful *Sound Forge* sample editor for Windows, Sonic Foundry is following up with a junior version, which works with most Windows sound cards. *Sound Forge XP* ($149) provides such sound-editing features as fade, pan, normalize, reverse, smooth, volume, time compression, and stereo-to-mono conversion. Effects include delay/echo, chorus, distortion, flange, and reverb.

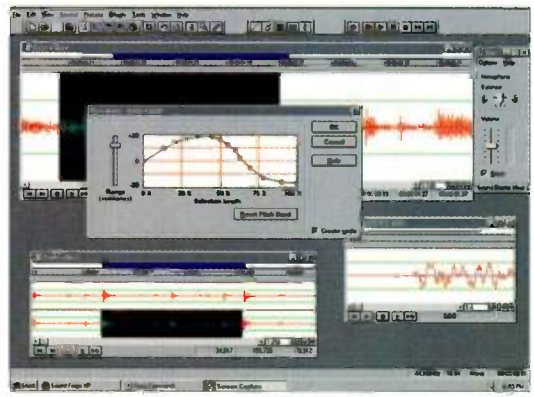
Like the "pro" version, *Sound Forge XP* supports a variety of audio file formats, including WAV, AIFF, VOC, VOX, and Sound Designer I. The program can open more than 50 files at once, each in its own window, and offers docking toolbars and pencil editing. The main features from *Sound Forge* 3.0 that have been omitted in *XP* are MIDI support, playlists, regions, and hooks for DSP plug-ins.

Speaking of DSP plug-ins, Sonic Foundry has announced its first *Sound Forge* 3.0 plug-in, *Noise Reduction* ($249), which analyzes and removes background noise such as tape hiss and electrical hum. The program creates a *noiseprint* from the background noise and uses it to distinguish the noise from the program material, allowing it to reduce noise with minimal timbral effect on the cleaned-up audio.

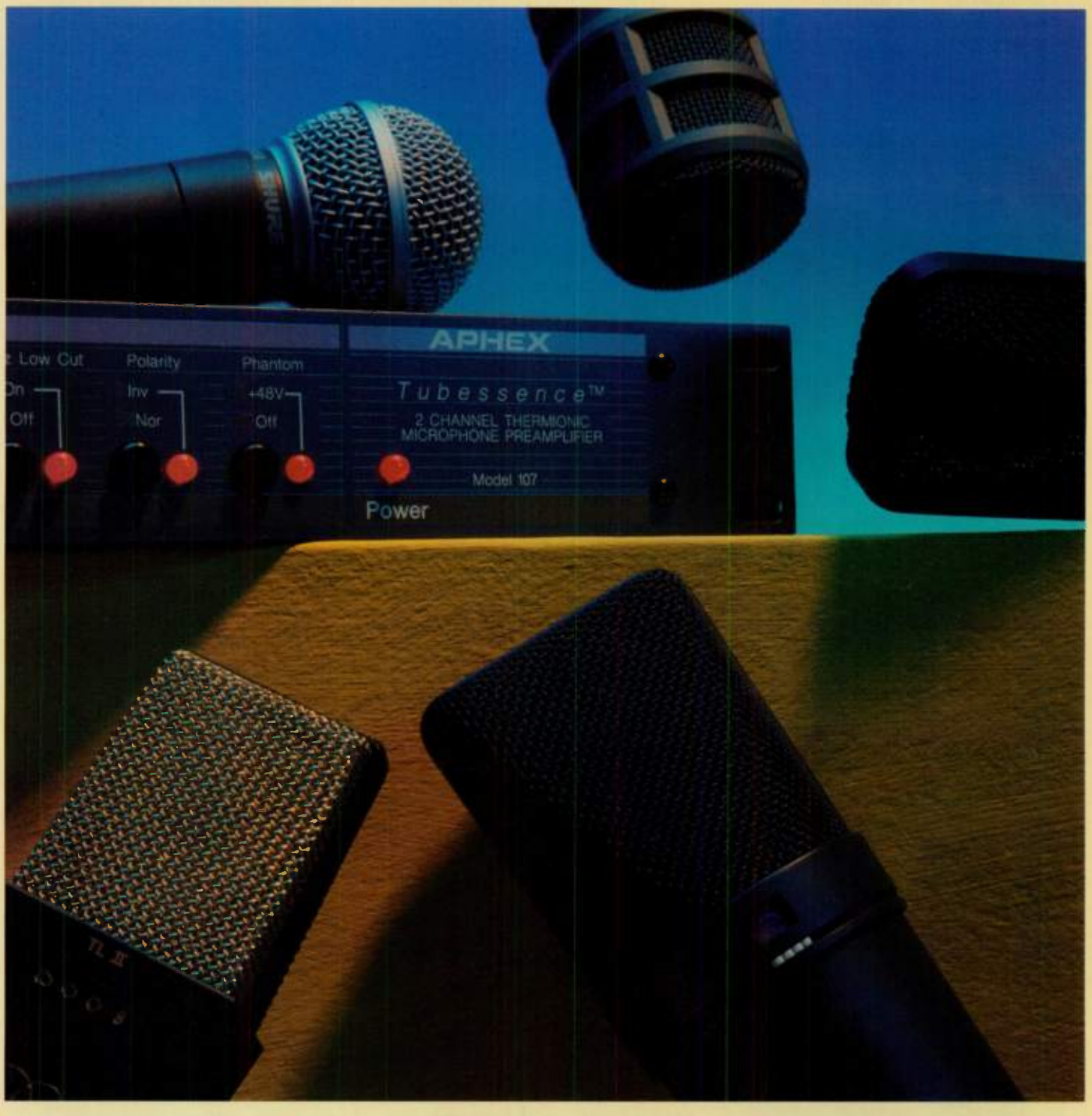
The plug-in comes with a Click Removal tool for eliminating clicks and pops, which is important when remastering old recordings. The garbage can be removed manually or automatically.

Although the Click Removal tool is part of the *Noise Reduction* package, the two functions are installed separately under Sound Forge's Tools menu. Sonic Foundry; tel. (800) 577-6642 or (608) 256-3133; fax (608) 256-7300; BBS (608) 256-6689.

Circle #408 on Reader Service Card



Upgrade all your microphones



Smooth and intimate, dimensional and detailed...is that the sound you're looking for? The Aphex 107 Tubessence® Thermionic Microphone Preamplifier reveals the subtlety and power in both vintage condensers and popular dynamic mics. Qualities that are lost on your console's mic preamp.

Tubessence, for the long journey from microphone to CD.

The Aphex Model 107, tube mic pre - two channels of Tubessence for $595.00 MSRP (U.S.). Call or fax for a dealer near you.

1995 TEC NOMINEE

APHEX SYSTEMS

See us at AES Booth #918

Improving the way the world soundsSM

11068 Randall Street, Sun Valley, CA 91352 • Tel: 818-767-2929, Fax: 818-767-2641

▶ DEMARIA LABS ADL 200

If you don't know that tube processors are "in," you must have been sleeping for the last few years. You see, Mr. or Ms. Van Winkle, the natural warmth of tube distortion often proves an excellent complement to the almost colorless accuracy of digital recording technology; it can improve your live sound, too. Direct-injection (DI) boxes offer a promising application of this classic technology, letting you warm up any instrument or line-level signal before recording or mixing it.

Anthony DeMaria Labs' ADL 200 ($799) is a serious 2-channel, all-tube DI box with individual ground lifts; ¼-inch inputs; XLR, mic-level outputs; and ¼-inch, line-level outputs. The ¼-inch outputs can be run direct or buffered through the tubes, while the XLR outputs always run through the tubes. It uses 12AX7 tubes, which are the most common type in audio preamps.

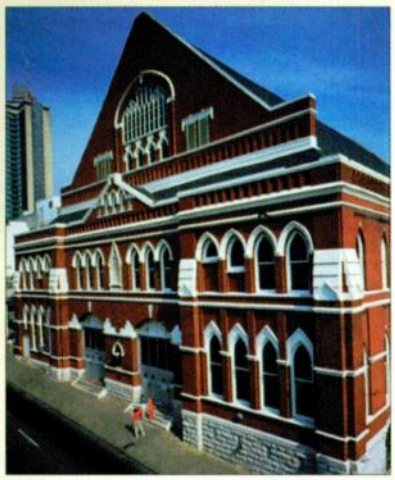
The ADL 200 is built into a heavy-duty chassis that can be placed on a tabletop or mounted in a half-rackspace. It includes a ¼W, fast-blow fuse. The manufacturer claims impressive audio specs, including a frequency response of 20 Hz to 100 kHz (±1 dB) and output noise at -85 dB. Anthony DeMaria Labs; tel. (818) 340-0228; fax (818) 340-4331.

Circle #409 on Reader Service Card

1995 NAMM SUMMER SESSION ▲ ▲ ▲ ▲ ▲

Even before a single product was unveiled, the Summer NAMM show in Nashville was hot. So hot, in fact, that I had to be coaxed out of the plane.

But although *I* could survive only by running between air-conditioned environments, Nashville is experiencing a tremendous boom. Musicians are moving to Music City in droves, and they're not exclusively country artists, either. Steve Winwood and Peter Frampton are just two of the latest rockers to call Nashville home. And according to local industry publicist Jimmy Miller, 40% of the music produced in Nashville recording studios is decidedly noncountry.



Nashville's Ryman Auditorium was the original home of the Grand Ole Opry. (Courtesy Gaylord Entertainment, Bill Lafevor.)

Fortunately, there were some *cool* things about the city: friendly people, amazing restaurants, and musicians with monstrous chops hanging out everywhere. All in all, the trip was a noteworthy experience.

Summer NAMM is smaller and way less frenetic than the gigantic Winter NAMM convention in Anaheim, California. However, that doesn't mean the summer show is a sleeper or a mere sideshow for guitar pickers. Electronic musicians were well served by some exciting product introductions.

Yamaha energized the show with the U.S. debut of its revolutionary 02R digital mixing console. (The 02R was previewed in our August 1995 "What's New" column.)

E-mu joined the modular digital multitrack revolution with DARWIN, a stand-alone, 8-track hard-disk recorder. The interface is as easy to use as an ADAT or DA-88, but DARWIN offers the benefits of random-access digital audio, and you can purchase the unit with an optional 1 GB hard drive for turnkey operation.

For a show that is often perceived as guitar-oriented, Summer NAMM was quite synth heavy. Korg introduced two sizzling new models, the Prophecy solo synth and the Trinity keyboard workstation (see sidebar "The Star of the Show"). Alesis clocked in with three dazzling premiers: the QS6 64-voice synth, the updated and improved MicroVerb 4, and the DM5 rack-mount drum module. And Roland premiered its XP-10, a truly fat-sounding, entry-level keyboard synth for a scary list price of $895.

For desktop recordists, the big news was probably the release of version 2.5 of OSC's *Deck II* hard-disk recording software for the Mac.

On the retro side of the fence, I was entranced by Groove Tubes' EQ1 tube equalizer ($595). The monaural, half-rack unit can be linked to another EQ1 for stereo processing.

The cool trend for performing artists was lower-priced in-ear monitor systems. Until recently, these systems were cost-prohibitive for anyone but superstars. However, CMCI is offering its Oracle Jr. system for just $989, and representatives from Garwood stated they are also developing a system for the under-$1,000 price point. Soon, club musicians should be able to benefit from the exquisite articulation of in-ear monitors.

Of course, this is just a little "flavor" of the show. Steve Oppenheimer, our marvelous gear guru, will present full reviews and detailed "What's New" announcements of these and other exciting products throughout the coming months.—*Michael Molenda*

NOW APPEARING
At A TASCAM Dealer Near You

DAT MASTER II

THE GREAT SOUNDING SEQUEL

INTRODUCING THE TASCAM DA-30 MKII

STARRING NEXT GENERATION DAT TECHNOLOGY WITH UNBEATABLE TASCAM QUALITY AND DURABILITY

INTRODUCING ADVANCED CONVERTER TECHNOLOGY FOR INCREDIBLE SOUND

FEATURING A DATA/SHUTTLE WHEEL FOR HIGH SPEED CUEING AND EXTREMELY FAST LOCATING AND PROGRAM ENTRY

BALANCED AND UNBALANCED ANALOG INPUT AND OUTPUT PLUS AES/EBU & S/PDIF DIGITAL I/O

AND A BUILT IN PARALLEL PORT FOR CONTROL I/O FROM EXTERNAL EQUIPMENT

AND IT'S ONLY $1,499*

IF YOU LIKED IT BEFORE, THE SEQUEL WILL BLOW YOUR MIND. THAT'S RIGHT. IT SURPASSES THE ORIGINAL DA-30. WITH AUTOMATIC CALIBRATION, MULTIPLE SAMPLE RATE SUPPORT, SCMS FREE WITH SELECTABLE COPY ID, AND LONG RECORD TIME FUNCTION FOR DOUBLING PLAYBACK TIME OF STANDARD TAPE, THE DA-30 MKII IS THE BEST VALUE IN A DAT MASTERING DECK ANYWHERE. IT'S PLAYING AT A DEALER NEAR YOU — GET THERE AND EXPERIENCE THE EXCITEMENT OF THE TASCAM DA-30 MKII.

TASCAM

Take advantage of our experience.

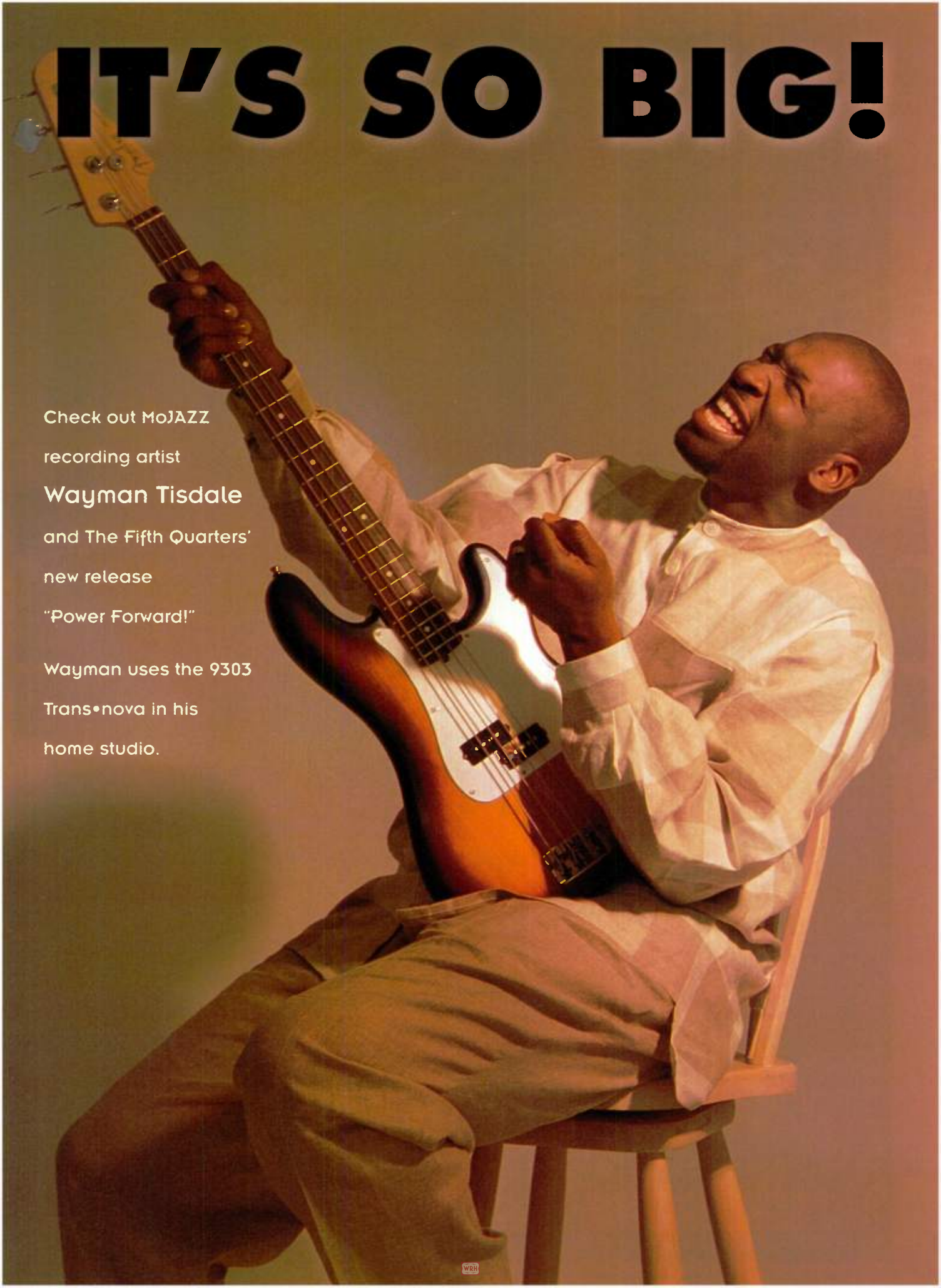
DAT The DA-30mkII is intended for mastering work; marks it as the successor to the original DA-30 which is the standard mastering DAT deck for production facilities worldwide.

See us at AES Booth #400

For information and specs via fax call 800-827-2268, and request document #2230

TASCAM FAXBACK

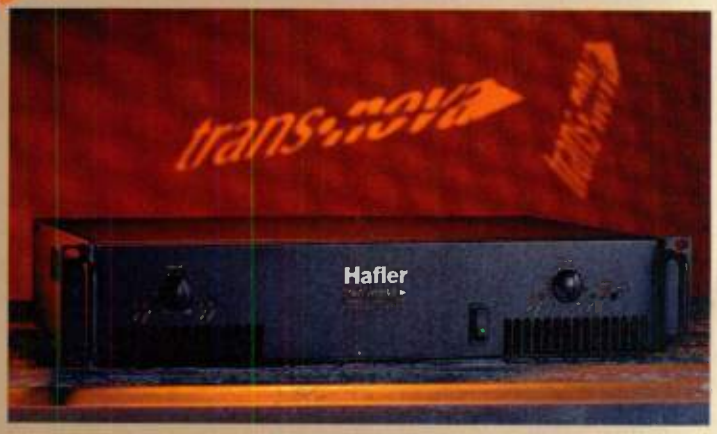
*Manufacturer's suggested retail price


IT'S SO BIG!
Check out MoJAZZ
recording artist
Wayman Tisdale
and The Fifth Quarters'
new release
"Power Forward!"
Wayman uses the 9303
Trans•nova in his
home studio.

Introducing the Biggest Hafler Ever!

Hafler
trans•nova
P7000 DIABLO

NEW!



Diablo

Dynamically Invariant A-B Linear Operation is an extremely important advance in front-end/driver circuitry for directly, exciting, large arrays of power MOSFETs. Traditionally, such circuits have only 6dB of surplus current headroom over their static or quiescent bias current. The new circuitry works on a variable-gain current-steering principle which results in absolutely linear current headroom of 20dB or more at low driver dissipation. This is extremely important for providing the high transient current needed to drive the capacitance of MOSFET gates at high frequencies. The result is both lower distortion and greater inherent stability.

Finally a Big Amp with Hafler's Legendary Sound Quality!

D.I.A.B.L.O. patent application in progress.
Trans•nova patent #4467288

- Sound Reinforcement
- Power Large Monitor Speakers
- Bass Guitar and other Instruments
- Sound Quality in a Big Amp!

P7000 Specifications

- EIA (1kHz, 0.1% THD)
- 375 Watts/Channel into 8Ω
- 525 Watts/Channel into 4Ω[A]
- 1050 Watts Bridged Mono into 8Ω[A]

[A]Continuous sine wave power limited by current rating of line fuse.

The award winning 9303 and 9505 Trans•nova amplifiers are already a huge success!

Hafler 7 YEAR WARRANTY

Hafler's big amps can be heard from Coast to Coast!

"The new Hafler Trans•nova D.I.A.B.L.O. sounds great here at Capitol's Tower Mastering. Dependable as always - clear and detailed with wonderful bass response."

Wally Traugot
Tower Mastering
Capitol Records
Hollywood, CA

"The D.I.A.B.L.O. circuitry makes an already great sounding amplifier even better. The high end is truly vacuum tube like, while the bottom has the punch of solid state.

The first project mixed with the new D.I.A.B.L.O. amplifiers is a Circle Surround DMP Big Band CD which includes classic charts from Basie, Dorsey, Ellington, Herman, Kenton and Miller."

Tom Jung
DMP Records
Stamford, CT

DMP's 5 channel Music Mixing Suite Powered by Hafler 9505 Trans•nova D.I.A.B.L.O. Amplifiers.

See us at AES Booth #1404

HAFLER PROFESSIONAL, A DIVISION OF ROCKFORD CORPORATION • TEMPE, ARIZONA 85281 USA
1-800-366-1619 • INTERNATIONAL FAX 602-967-8132 • IN CANADA, CONTACT CABLE TEK 604-942-1001

MADE IN THE USA

THE STAR OF THE SHOW ▲ ▲ ▲ ▲ ▲

KORG TRINITY

Keyboard workstations—keyboard synths with onboard sequencers and effects—are the workhorses of many MIDI studios. With this in mind, Korg offers four versions of the Trinity, its latest sample-based keyboard workstation. The new series, which should be available late this year, includes the 61-key Trinity ($3,599), enhanced 61-key Trinity Plus ($3,999), 76-key Trinity Pro ($4,799), and weighted 88-key Trinity ProX (price tba). All are sensitive to Velocity and Channel Pressure.

All four models include a 3.5-inch HD floppy drive, an internal power supply, four analog outputs, and a 60,000-note, 10-song, 16-track sequencer with 192 ppqn resolution. The sequencer reads and writes Standard MIDI Files.

The Trinity's user interface is the most immediately noticeable new feature, especially the large, 320 × 240-dot, graphic touch screen called TouchView. When used with the onboard sequencer, for example, this feature lets you manipulate track sliders and parameter values by touching the screen with your fingertip. An alpha wheel, data-entry slider, and increment/decrement buttons also help navigate through the software pages and set values. Rear-panel inputs accept a sustain pedal, a continuous pedal, and an assignable footswitch.

Real-time modulation sources include a pressure-sensitive (XZ) ribbon controller and Korg's familiar XY joystick. A pair of buttons located above the joystick can be programmed to send any MIDI switch message. Other control sources include the various envelope generators and LFOs, Velocity, Pressure, and more. A volume slider also is provided.

The Trinity uses Korg's Advanced Control Combined Synthesis System (ACCESS), a new DSP-based tone-generation technology that features PCM samples recorded and played back at 48 kHz. The unit's 256 Programs and 256 Combinations use a total of 24 MB of ROM samples (375 new multisounds and 258 drum samples). If the oscillators are not layered, the system offers up to 32-voice polyphony and 16-part multitimbral operation.

ACCESS includes a resonant, multimode filter that can be configured as lowpass, highpass, bandpass, or bandreject. Dedicated envelope generators are provided for the oscillator, filter, and amplifier, and dedicated LFOs modulate the oscillator and filter.

In Sequence and Combination modes, each Timbre (eight of which make up a Combination) can have its own insert effect, for a total of up to eight simultaneous effects, depending on effect size. You can also add overall ambience with two types of master effects. The effects processor provides over 100 effects, including basic reverb and delay, gating, distortion, overdrive, and rotary-speaker simulator.

In addition to the ACCESS sounds, all Trinity models except the basic 61-key unit include a second Solo Synth sound source that uses Korg's Multi Oscillator Synthesis System (MOSS), which is optional on the basic Trinity. The Solo Synth comes with 64 user Program locations.

The Solo Synth is a DSP-based, monophonic sound source designed primarily for solo parts. It combines physical modeling, analog, and Variable Phase Modulation (VPM) synthesis techniques, any two of which can be used simultaneously. MOSS provides two oscillators, a suboscillator, and a noise generator, which are internally submixed and sent to a resonant multimode filter and amplifier.

In addition to dedicated pitch and amplitude EGs, MOSS includes four assignable, general-purpose EGs and four assignable LFOs, which can generate a wide variety of waveforms. To top it off, you get ring modulation, cross modulation, hard sync, and two waveshaping sections.

The Trinity series will be available with several interesting options, all but one of which are included on the ProX. (Option prices have not been announced.) The exception is an 8 MB Flash RAM board, which lets you import Korg- or Akai-format samples and adds memory locations for 256 Programs and 256 Combinations. If you have the Solo Synth, the Flash RAM option also gives you 64 more Solo Synth Programs.

The optional HD REC board provides two simultaneous tracks of 48 kHz, 16-bit hard-disk recording and four tracks of playback with automated levels and panning, MTC sync, and S/PDIF digital I/O. An ADAT-format, optical digital interface with a word-clock input also is standard on the ProX and optional for the other models. In addition, the ProX comes with a 540 MB internal hard drive.

Korg also introduced the Prophecy ($1,599), a MOSS-based mono lead synth that includes essentially the same features as the Trinity's Solo Synth. The Prophecy has a 37-key, Velocity- and Pressure-sensitive keyboard; a ribbon controller; and an arpeggiator. The ribbon controller is mounted on a cylindrical device that can be rotated, providing another control axis. Five front-panel parameter knobs can be assigned to MIDI controllers.

The Prophecy's onboard effects processor offers seven algorithms, including distortion, wah, chorus, delay, reverb, autopan, and 2-band parametric EQ. Korg USA; tel. (516) 333-9100; fax (516) 333-9108.

Circle #410 on Reader Service Card

Studio Vision Pro 3.0

"The hands-down winner of the Towering Codehead Spectacular Achievement Award are the good folks at Opcode for Studio Vision Pro 3.0.

Say you had a lead vocal with a couple of flat notes. Convert the data to MIDI, and the flat notes will be represented as pitch-bend data. Edit the data, re-convert it, and lo and behold: perfect vocal!"

—Keyboard Magazine, May 1995

Perfecting a flawed vocal is only one of the feats that Studio Vision Pro 3.0 makes possible. Pro 3.0's built-in DSP, and Audio-to-MIDI™ conversion may just be the tools that the genius in you longs for. So contact us today and we'll show you how Pro 3.0 gives you the power and flexibility to convert your ideas into music.

Studio Vision Pro 3.0 for Macintosh, the next evolution of digital audio

Call **(800) 557-2633** ext. 241 for literature **(415) 856-3333** for dealer info

3950 Fabian Way, Suite 100 • Palo Alto, CA 94303 • Fax: (415) 856-3332

OPCODE

your music first™

World Wide Web: http://www.opcode.com

HIGH TECHNOLOGY
WITHOUT THE
HIGH ANXIETY

STRUM A SAX. TONGUE A DRUM. HIT

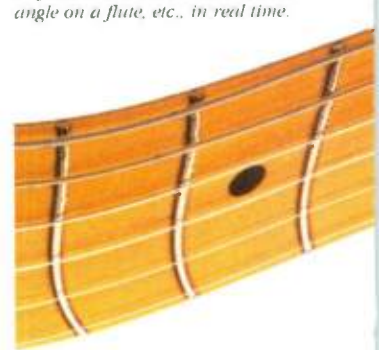
Mold and shape up to 256 sounds and 128 combinations per bank.

Maximum of 64-note polyphony gives you full MIDI orchestration capability.

64 NOTE POLYPHONY

IT JUST MAY BE THE MOST SIGNIFICANT BREAKTHROUGH IN SOUND SYNTHESIS TECHNOLOGY. INTRODUCING THE TECHNICS ACOUSTIC MODELING SYNTHESIZER. WITH THIS AMAZING INSTRUMENT, YOU'LL ACTUALLY CREATE NEW MUSICAL SOUNDS BY COMBINING THE DRIVER OF ONE INSTRUMENT -

Control where you strike the drum surface, determine the mouthpiece angle on a flute, etc., in real time.



A TRUMPET. AND BLOW YOUR MIND.

Real-time expression lets you control sounds as you would physically, such as bending strings.

SUCH AS A GUITAR PICK - WITH THE RESONATOR OF ANOTHER - THE TUBING OF A SAX. YOU'LL EVEN INVENT INSTRUMENTS THAT CAN'T EXIST IN REALITY. THE NEW TECHNICS ACOUSTIC MODELING SYNTHESIZER. IT WILL CHANGE THE WAY YOU CREATE MUSIC FOREVER.

Imagine creating your own instruments. How does a saxadrum sound?

Technics®
The science of sound

Proud Sponsor of the 1996 U.S. Olympic Team

SOUND THINKING ▲ ▲ ▲ ▲ ▲

▲ NORTHSTAR PRODUCTIONS

Phase 5 ($695), Northstar's fifth CD-ROM of assorted instrument sounds for the E-mu EIII/ESI/EIV sampler family, is jam-packed. It delivers 91 Banks and 2,900 Presets, and Bank sizes range from 2 to 32 MB.

One highlight of the disc is its 1,680 jazz and pop electric guitar chords, each of which is played in multiple fret positions on a Strat-type guitar. Each chord is sampled with a slow downstroke/upstroke; short, hard downstroke; and quick upstroke with a chord mute.

Brass and saxes also are featured, including trumpet, trombone, and sax sections; cup- and Harmon-mute trumpet; comedy trumpet; and tenor and soprano sax. Completing the volume is a selection of jazz basses, Hammond organs (with and without Leslie), acoustic and electric pianos, vibes, congas, full jazz brush drum kits, and group vocal sections.

A "lite" version ($199) contains mono versions in which the Bank sizes are reduced. It comes with a $200 coupon toward upgrading to the full version.

Also from Northstar is the *New Gold 3* sampling CD-ROM ($595; $845 on 650 MB magneto-optical disk) for the Akai S1000 family, which appears to be a 24-karat source for ethnic string, wind, and percussion samples. Stringed instruments include sitar, Renaissance lute, psaltry, banjo, zither, and assorted Chinese kotos. The "things people blow" category includes Egyptian Arghul, Syrian Midjwiz, Arabian Mizmars, and Tibetan Mountain Pipe.

Percussion sounds from Latin America and India join Native American drums and Javanese gamelans. African percussion includes Sugu, Chueshee, Gome, Odunu, and Ashaka/Ngu Ngu.

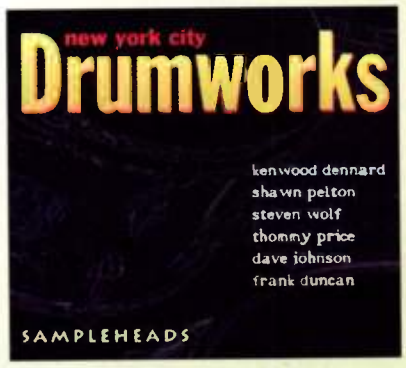
To top it off, Northstar includes sixteen sound effects, ranging from "Yellowstone Geysers" to "New York City Traffic." *New Gold 3* includes more than 2,200 Programs and more than 248 MB of samples. Northstar Productions; tel. (503) 760-7777; fax (503) 760-4342.

Circle #411 on Reader Service Card

▼ SAMPLEHEADS

What if six top New York session drummers—say, Kenwood Dennard, Shawn Pelton, Steven Wolf, Thommy Price, Dave Johnson, and Frank Duncan—decided your studio was a hip place to lay down some tracks? If your production chops equal those at Sampleheads, it might sound like *New York City Drumworks* ($99; CD-ROMs $249), the company's audio CD sampling library.

The 2-CD set holds 146 minutes of rhythms by the aforementioned stickmeisters, including funk, rock, hip hop, Latin, Cajun, R&B, Brazilian, rockabilly, African, reggae, swing, gospel, and country beats. The prelooped parts were performed at several tempos, using sticks, brushes, and mallets. The 1,252 events, 735 loops, and 517 hits are copyright free. A booklet provides index and timing information.



East-West Communications (distributor); tel. (310) 858-8797; fax (310) 858-8795.

Circle #412 on Reader Service Card

KID NEPRO PRODUCTIONS

Kid Nepro Productions has created five new sound banks for the Korg 01W and 03R/W series ($40/set): Synth Mix, Acoustic Set, Soundtrack/Film Mix, Mixed Bag, and Techno Dance. Each set includes 100 Programs and 100 Combis.

KNP has also released ten more sound banks for the Roland JD-990 ($40/set). Six of these—House Mix, Classic Synths, Pop Mix, Soundtrack, New Age/R&B, and Rock Collection—are based on the synthesizer's onboard waveforms. The Pop Ex Sounds, Orchestral Ex Sounds, Vintage Ex Sounds, and World Ex Sounds banks rely on waveforms from Roland expansion boards. The Ex sounds are also compatible with the JV-80/880 synths.

All sound sets are available on floppy disk in Atari, Mac, MS-DOS, and Alesis DataDisk format. Kid Nepro Productions; tel. (718) 642-7802; fax (718) 642-8385; e-mail kid nepro@aol.com.

Circle #413 on Reader Service Card

PATCHMAN MUSIC

Roland's JV-80/880 has proven to be a favorite of programmers in recent years. Patchman Music has added its vote of approval with two new sound banks for this Roland mainstay. Each sound bank ($29.95) includes 64 Patches, sixteen Performances, and one GM Rhythm Setup. The sounds are available in Mac self-loading or Opcode *Galaxy* format, or as Macintosh or PC Standard MIDI Files.

Volume 1 fills in some fundamental sounds that are missing in the JV-80 or that Patchman feels were poorly done. Volume 2 emulates classic analog synth leads, basses, pads, brasses, and strings. Patchman Music; tel. (216) 221-8887; e-mail matteblack@aol.com. ●

Circle #414 on Reader Service Card


MTV
MUSIC TELEVISION®
GL4
MTV Europe and Allen & Heath
An association of technical excellence
GL4
Everyday, audio professionals everywhere are finding the GL4 superior in quality of audio path, ease of operation, and for flexibility in front of house and monitor applications, and overall value. Contact your professional audio dealer today and audition the console that's raising eyebrows around the globe.
ALLEN & HEATH
A Harman International Company
See us at AES Booth #606
Allen & Heath: 8760 South Sandy Parkway • Sandy, UT 84070 • (801) 568-7660 • Fax (801) 568-7662

PRO FILE

Fusion Zone

Jacky Schreiber tells tall tales with rhythm.

By Diane Lowery

Imagine combining the music of Stockhausen, Zappa, Bach, and Gong. Stir them together and you have the music of Jacky Schreiber. For Schreiber, it was natural to combine all these influences into his independent CD release, *Zona de Fusion*. The result is a complex, fascinating blend of textures and rhythms. Weaving together rhythms from Venezuelan, Arabic, Egyptian, and Indian folk music, Schreiber fuses diverse sonic images into a truly unique musical style.

"I want to tell a tale with each song," says Schreiber, "but I want the stories to unfold in several layers, as if many different people were reading them at the same time. To accomplish this, I mix rhythms together to produce different feels, or 'stories.' I must often create hundreds of mixes for a single song until I discover the best way to converge all these rhythmic stories."

Schreiber's extremely intricate use of rhythmic textures stems from his belief that rhythm is more than just "groove." He conceptualizes rhythms as melodies and harmonies that create distinct musical landscapes. If the melody of a song changes, he'll often change the rhythmic "melody" to match.

"I think of rhythm as a state that is ever changing," he explains, "because rhythm is the main ingredient of not only music but of everything in the universe. Rhythm is present in harmony, melody, and even outside phenomena, such as your breathing or your heart beating."

Although Schreiber composes at a piano, he records with two main tools: Ensoniq's EPS-16 Plus sampler and Dr. T's *KCS* sequencing software. He programs basic sequences in real time—usually starting with a simple pattern or sound—and then improvises over them, developing countless variations on the original theme.

"The song 'Guitra' is a fine example of my process," says Schreiber. "I started with a 7/4 rhythm using a simple arpeggio and a bell-like sound. I worked out the song structure from start to finish with only those elements. Then, I wrote a series of other contrapuntal elements that danced around the central rhythm."

So how did Schreiber layer all his rhythms and polyrhythms so that *Zona de Fusion* didn't sound muddy? "I usually record all my instruments dry and process them with a very subtle reverb at the mixdown. Occasionally, I will record an instrument direct with reverb, but I take into account that I can't remove the reverb later. As a producer, one has to be careful of painting himself or herself into a corner.

"I also make use of natural decay and ambience when recording acoustic instruments. I think one secret of producing clean mixes is to let the sound and rhythm of each instrument create its own effect."

Schreiber has let music from all over the world drive his creative muse. "It doesn't matter whether you are composing, playing an instrument, producing, or whatever," he says. "The medium in which you are working drives the engine of creation." For Schreiber, rhythm is the engine that creates his tales of complexity and beauty.

For more information, contact Jacky Schreiber, Av. "A", Qta. Eva, P.B., La Carlota, Caracas 1071, Venezuela.

Diane Lowery is the copy editor of **EM**.

Jacky Schreiber

Purchase In '95 And Receive 6-Year Warranty!

Totally Cool.

PowerLight Amplifiers from QSC are cool to the core.

They're possibly the best sounding amps that QSC makes. *Exactly what you want from your power source:* wide, dynamic range with crystalline highs, full, round mid-range and a very fat but clean low end. PowerLight amps are extremely efficient. They'll drive your speakers better than they've been driven before. Even at 2 ohm loads.

PowerLight Amps also sound great because of *what you won't hear* —rattle and hum. These amps have remarkably low distortion due to improved output circuitry. And the noise typically associated with transformers is completely absent.

These are the only light amps that won't interfere with your wireless mics. *No static and no loss of radio signal.* PowerLight Amps meet all RFI and FCC requirements in the U.S. and EMC requirements in Europe.

PowerLight Amps run very cool, even when the performance is very hot...even in the middle of summer. The high aspect ratio aluminum heat sink was designed with the maximum possible surface area to absorb and dissipate heat aggressively. Variable speed fans blow it away on demand. (And signal muting kicks in when things get extreme to prevent shutdown.)

PowerLight Amplifiers from QSC—the best sounding, most advanced package of power technology available. And, at a cool 18 pounds, you can put them wherever you need them without sweat. And that's cool. So, pick some up from your QSC Dealer. When you purchase throughout 1995, you'll receive QSC's extended warranty at no additional cost, for a total of six years coverage.* Call QSC for more information at 714-754-6175.

LOAD IMPEDENCE	PowerLight 1.0	PowerLight 1.4	PowerLight 1.8
2 OHM*	500 Watts	700 Watts	900 Watts
4 OHM**	300 Watts	500 Watts	700 Watts

*1 kHz, 1% THD, TYPICAL; **20 Hz to 20 kHz, 0.1% THD

QSC® AUDIO

*If you purchase PowerLight Amplifiers by 12/31/95 you will receive QSC's 3-Year Extended Warranty in addition to the 3-Year Standard Warranty.

PowerLight is a trademark and QSC is a registered trademark of
QSC Audio Products, Inc. 1675 MacArthur Blvd., Costa Mesa, CA 92626-1468
(714) 754-6175 FAX (714) 754-6174

See us at AES Booth #151
Demo Room #DO3

AUDIO VISIONARY

SUPER-PRODUCER TONY VISCONTI TALKS ABOUT BOLAN, BOWIE, AND THE BENEFITS OF WORKING AT HOME.

BY MICHAEL MOLENDA

Tony Visconti has worked with just about every legend, singer/songwriter, and scoundrel in rock. Although he is best known for his classic works with David Bowie and T. Rex, his audio productions have made a lasting imprint on every decade since the late 1960s. Visconti's credits, therefore, are almost impossible to list comprehensively in the print medium—we'd need to dedicate an entire issue to his discography alone! An (extremely) short list of luminaries he has directed from behind the control-room glass includes Adam Ant, The Alarm, Badfinger, Boomtown Rats, John Hiatt, Mary Hopkin, Paul McCartney (as orchestrator), Moody Blues, Tom Paxton, Iggy Pop, Thin Lizzy, and U2.

Even more amazing than Visconti's prolific output, however, is the fact that most of his productions are truly pioneering achievements in the use of signal processing and manipulation of the audio soundstage. Not bad for a kid who started out playing the Catskills!

The Brooklyn-bred Visconti actually did begin his career

ascent playing double bass in jazz clubs and Catskill resorts when he was only sixteen years old. He soon focused on becoming a songwriter and recording artist, that is, until his music publisher recognized the quality of Visconti's song demos. In 1967, the publisher "loaned" Visconti to legendary British producer Denny Cordell and instructed him to report back on how the English made records. The spy mission was supposed to last six months. It ended up being a 23-year assignment, and during that time, Visconti produced so many classic tracks that the British (like everyone else) were stealing *his* riffs.

But even though Visconti has produced some of rock's biggest stars and worked in some very heavy studios (including his own Good Earth, a world-class facility in London, from 1972 to 1989), he has also embraced the personal studio. In his current home in New York, he has installed an ADAT and Mac-based studio with myriad digital editing tools.

As a consequence, Visconti has developed a production methodology that affords him the luxury of tracking in large studios and doing editing and overdubs in the comforts of home. And Visconti is no less an audio pioneer within the constraints of the "small" studio; he has developed all kinds of tricks for making his home-grown digital tracks sound as sweet and sexy as the analog productions that made him an industry legend. On occasion, he will also share his tips, tricks, and experience with America Online subscribers in the Producer's Forum of Composer's Coffeehouse (keyword: composers). Visconti is definitely a sage for anyone who is considering a career as a producer.

PRODUCER TALK

Originally, this feature was planned as a conventional prose interview. However, Visconti's responses were so packed with information (and some wonderful anecdotes) that I felt it would be criminal to rob **EM** readers of the complete dialog; hence the question and answer format.

I tried to organize the interview to run somewhat chronologically through the main aspects of the producer's craft: finding talent, critiquing and arranging material, eliciting passionate performances, optimizing technological tools, and recognizing the value of unexpected "gifts." Visconti's responses to these questions almost form a mini-textbook on the producer's art, offering a blow-by-blow account of what happens when an artist and producer are locked into their creative fertility dance.

LARRY BARNS

Tony Visconti's home studio gives him the flexibility to undertake "labors of love," such as producing an album for singer/songwriter Alex Forbes.

What elements must an artist possess to seduce you into wanting to produce him or her?

That's like asking, "How do you choose your spouse?" The answer is I fall in love. However, a recurring theme in my work is a quirky, unique voice. When I first heard Marc Bolan [of T. Rex] singing in a small London club, I was bowled over by the quivering, Delta blues-inspired quality of his voice. It was unearthly, and I just had to move closer to find out where that sound was coming from. His delivery was almost feminine, and I couldn't decipher the words or even confirm whether he was singing in English. I was completely enthralled by his voice.

The quality of the artist's material is also very critical. The songs must sound like they could be "classic" works while also offering something that is new and unusual. Perhaps not coincidentally, most of the truly successful artists I've produced wrote their own songs. Bolan didn't have instant chart success, but he had a unique songwriting style that I could help develop. When we hit it, we had a continuous run of success for three years and formed a style that is still being copied today.

Finally, I want to be sure an artist has chops. I don't require that they be classically trained or schooled, but they must have talent up the wazoo and a great sense of how to project that talent. Although I can perform all sorts of sonic miracles in the studio, it's more exciting to work with a great musician than to throw a zillion vocal takes into a computer and "fix" all the out-of-tune notes. A greater form of magic is still the traditional kind: where an artist stands in front of a mic and sings his or her heart out.

Is there any way to gauge whether a working relationship will be productive or not?

After I hear the artist's demo and see him or her perform live, I just follow my gut feeling and go with it. At that point, there's no way to know what will happen. But when we actually start working together, I try to foresee as much as possible what level of sanity I'm going to have to deal with in the studio. It is my experience that all artists are either insane or big babies; it just has to be gauged to what degree they are afflicted. Of course, I'm being facetious here, but the "honeymoon" usually comes to an end after the first week in the studio, and then you see what the artist is *really* like. It can be quite scary sometimes.

Let's say an artist isn't delivering the level of performance that you believe he or she is capable of, what do you do?

I will do anything that comes to mind, including some real disgusting stuff.

MIXDOWN

TAKE CONTROL AND DISCOVER NEW FRONTIERS

CLASSIC 8

MIXDOWN CLASSIC 8 - THE LATEST GENERATION OF PROFESSIONAL AUDIO CONSOLES FOR MULTITRACK RECORDING OR LIVE SOUND REINFORCEMENT

Launched in early '88 the Mixdown was followed two years later by the Mixdown Gold with extra features and upgraded spec. Both were critically acclaimed everywhere they landed. That was then. We've now smashed through to a new dimension with the Mixdown Classic 8 - the latest generation of professional audio consoles designed to face the future. The MIXDOWN CLASSIC 8 offers unsurpassed quality, features and specifications that a few years ago would have been unthinkable at the price! With three models in the range 16, 24 & 32 channel, and with the full width meter bridge, MIDI muting and rackmount EP5 power supply - you're entering into a whole new world of music making.

METER BRIDGE - The adjustable meter bridge has a two colour, 12 segment peak reading LED display for every channel and moving coil VU reading meters for the left and right outputs.
INPUT CHANNELS - All input channels are crammed full of features including balanced mic and line inputs, inserts, direct outs, +48V. 20dB mic pad, three band EQ with sweep Mid sweep Low & EQ cut, six aux busses, MIDI mute, PFL and smooth 100mm fader.
OUTPUT SECTION - The sixteen tape monitors can be used as extra inputs bringing the total number on a 16 channel up to 34. The upper row of inputs even feature two band EQ, PFL, a couple of aux sends and fader reverse. All output groups have insert points. 12 segment bargraphs and 100mm faders. A line up oscillator, stereo return and a built-in talkback mic.

1995 TEC NOMINEE

WHEN YOU NEED TO GET SERIOUS....

Studiomaster

Studiomaster Inc. 3941 Miraloma Avenue, Anaheim CA92807 USA Tel 714 524 2227 Fax 714 524 5096
Studiomaster House, Chaul End Lane, Luton, Beds, LU4 8EZ UK Tel 01582 570370 Fax 01582 494343

I learned years ago, while playing improvisational music, that the music one makes is a result of one's experiences. So if artists are stuck, I often give them an "experience." This might take the form of a practical joke, or perhaps we'll just sit down and drink a little wine. I've even been known to employ a strip-o-gram to get a band to loosen up. The process of making music should be both passionate and fun. If those two vital ingredients are missing, it's not worth being in the studio.

It's also important that the artists feel comfortable—young bands, in particular, often find studios to be very intimidating—so I do my best to cloak the studio's high-tech surroundings. I might bring in table lamps to replace harsh lighting fixtures, and I encourage groups to post their favorite works of art and other familiar items around the studio. Cameras are a great source of fun, too. If you pull out a Polaroid or a Handycam, instant mirth forms in the studio.

If a performance still isn't happening, I call a break and sit with the artist to talk about the "meaning" of the piece. Sometimes musicians can get too focused on details and forget the big picture. I think it's important that the artist be reminded of what he or she is trying to communicate with music. For example, the song isn't about singing the third chorus louder than the second, it's about a woman who has left you for a trapeze artist and broke your guitar before she left with your credit cards!

Also, it's not the end of the world if you don't get the vocal or solo during the scheduled session. If the artist's biorhythms aren't right, we simply take a break and try again later. I keep a lot of tracks free, because I like to use at least eight separate tracks for lead vocals and solos. I'm a firm believer in keeping everything the artist puts on tape and then editing the best performances together on a composite track; if I'm out of tracks on the master tape, I bounce the backing tracks to a slave reel and continue recording. I like to save everything, because I've discovered that everything tends to sound much better after a day's rest. I find that the little

TONY VISCONTI

Flower children, cosmic dancers, and founding members of T. Rex: Steve Peregrine Took (left) and Marc Bolan.

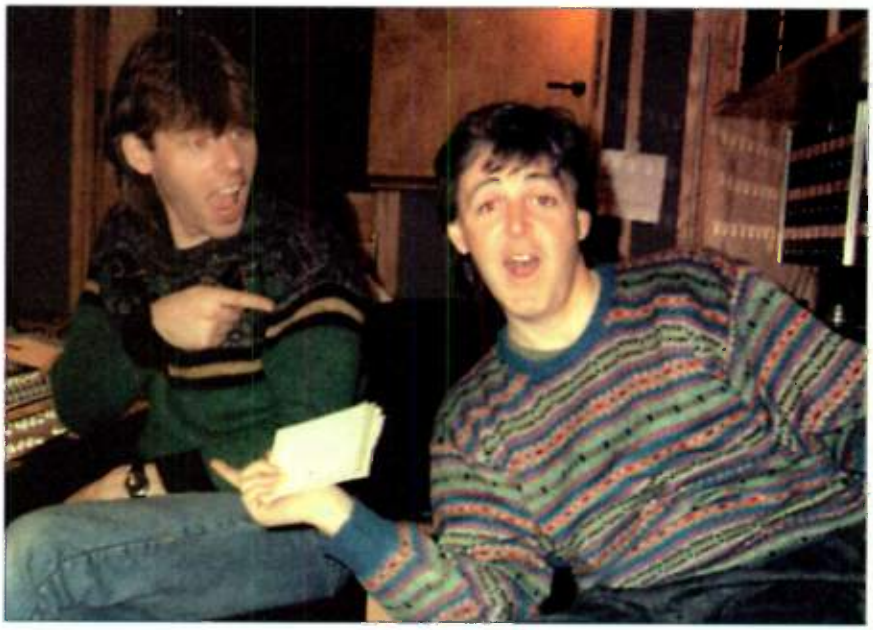
mistakes that irked me are forgotten and that the overall performance was far better than I had imagined.

How much do you involve yourself in reworking an artist's material?

Oh, I get involved! I hate to see a basically good song go by the wayside due to a glaring fault. I consider myself a "song doctor." I try not to actually write anything myself—I know that there are many producers who take a writing credit just for changing a few notes—opting instead to help the writer take the song one step further towards being a classic song. I think it's my job to do that.

So is there some secret pop producer's trick to making a song more commercial?

I just follow some basic rules, such as ensuring the song has a strong chorus that is heard several times. But let me give you a practical example: Bono phoned me after U2 released *Unforgettable Fire* and said that "A Sort of Homecoming" was picked for a single but that Brian Eno and Daniel Lanois couldn't edit down the 6-minute album version. The edits just didn't flow satisfactorily.



TONY VISCONTI

Producer Hugh Padgham (left) and Paul McCartney horsing around in McCartney's private studio while recording *Press to Play*. Visconti orchestrated "Only Love Remains" for the ex-Beatle.

I listened to the album cut and determined that commercial elements were already present but were not highlighted or repeated. A lot of attention was paid to very floaty, washy instrumental passages. I called Bono back

and suggested the band ignore the album format completely and define the main musical elements of the verse, chorus, and bridge.

Fortunately, the band was performing the song on tour and the live version was considerably different from the album version. We decided to re-record the song live and use that version for the single. So I rehearsed them during the sound check for a concert at Wembley, and they all loved the tighter arrangement I suggested. The original idea to use one of the subsequent concert performances was rejected because stage adrenaline prevented us from capturing a decent, well-played track. A recording made during the soundcheck rehearsal was used instead. We did some overdubs in the studio the following week, and when the new single version was released—on a 4-song EP entitled *Wide Awake in America*—it became a big hit.

Do you already have a strong sense of a song's musical arrangement during preproduction, or do you develop ideas during the overdub and mixing processes?

I have a good imagination, so I often "hear" the finished work before the rhythm track is even conceived. Most of the time I go for a strong rhythm track and make sure there is enough sonic space to overlay my ideas for strings, brass, or backing vocals. A great signature line during the song's intro that is repeated between the choruses and verses is just as important as the song itself. The arrangement is everything.

Having said that, Bowie usually managed to make arranging very difficult, because he'd seldom bring a finished lyric or melody to the sessions. He'd only have a basic idea, based on a book he'd just read or a recent conversation. Even the key of the song would be arbitrary. For example, the song "Fashion" was conceived as nothing more than a riff while the band was rehearsing in a house in Jamaica. The song was called "Jamaica" even after all the music was recorded! All the little musical "tastes" were recorded simply because they sounded good; they weren't embellishing the vocal melody, because there wasn't one. Months later, back in London, Bowie admitted he couldn't come up with a lyric or melody line and suggested we abandon the track. I vaguely remember pleading with him to come up with something, and the next afternoon he arrived with most of the song finished. Some lines were written on the spot as we recorded the vocal. Bowie is the only artist I've worked with who actually writes on mic!

Because you develop your arrangement ideas very early in the production process, do you tend to "stick to the plan" as the recording progresses?

I follow my vision but not the letter of my law. It's important to avoid being a slave to a concept that should really be under constant revision. This is because I think it's important that every musician be allowed to make his or her personal contribution to the group. I have no problem modifying my plans to accommodate any spontaneous magic that occurs when the musicians start playing.

At what point do you begin the sweetening process, and how do you decide which sonic elements will best enhance the track?

Almost immediately after a rhythm track is recorded, I begin laying down some elementary sweetening and don't stop until the basic track sounds great. I may ask the guitarist to double or harmonize some lines, or I'll request that the keyboardist double the bass line at certain points. Of course, there are always a few spots for the drummer to slap on a tambourine or shaker. These quick "tastes" can polish up a rhythm track without muting its punch and spirit. Another good reason to start the sweetening process fairly early is that these sonic enhancements can magnify whether the track is truly hot or not. If the track doesn't come alive after sweetening, we re-record the basic tracks, armed with a better idea of what must be changed.

The real intricate sweetening, however, takes place after all the rhythm tracks are finished. Then, we can break down the board from recording basics and plug in a whole bunch of toys, such as samplers, multi-effects processors, and various types of microphones. I don't mind spending all day getting a 30-second "eargasm" on tape, but I must have all the basics completed in order to be free to orient my mind solely towards the sweetening process.

How do I choose the actual sweetening elements? I adapt what works best with the song. The same riff can sound great on many different instruments,

ED SCIAKY

Bruce Springsteen (seated, left), David Bowie (standing, far right), and Visconti (seated, right) listening to the Thin White Duke's version of Springsteen's "Saint in the City" during the recording of *Young Americans* in 1974. Sadly, the track was never finished.

If You Need More Power Than This...

Maybe You Should Run For President

Our new MFA Series of amplifiers deliver the power that digital audio sources demand. Whether used for motion picture sound tracks or live contemporary music, these models faithfully reproduce the extended dynamic range of today's technology. High power, great sound, and a full Five Year Worry Free Warranty. Why buy anything else?

* 24,000 watts as illustrated: Eight MFA-8000 amplifiers combined, each putting out 1500 watts per channel into a 2 ohm load. Also available: the MFA-6000, with up to 900 watts per channel.

POWER AMPLIFIERS BY ASHLY

Ashly Audio Inc., 100 Fernwood Ave, Rochester, NY 14621 • Toll Free: 800-828-6308 • FAX: 716-266-4589 • Canada: Gerraudio Dist. Inc. 416-696-2779
International Agent: E and E Exports Inc. 714-440-0760 • Internet: http://www.ashly.com

and with MIDI modules the range of options is incredible. I trust my gut feelings to narrow the choices down. There are always emotive guidelines you can follow to select sounds that empathize with what the artist is communicating: guitars make you feel angry or passionate, brass makes you feel bold and cocky, strings make you feel sad or romantic, and so on.

So what inspired you to sweeten all those T. Rex tracks with classical orchestrations?

I was trained in music and Marc wasn't, but even though he knew only seven chords, he used to delight in anything I'd show him from the classical world. One of our favorite records was *Instruments of the Orchestra* narrated by Sir Adrian Boult, and if Marc heard something like a Cor Anglais, he would giggle with delight and say he wanted one on his records. So after "Ride a White Swan" became a hit, we made an agreement that I could suggest and write for any combination of classical instruments that I saw fit to be on a T. Rex record. Now, T. Rex records were made very quickly with no thought whatsoever of orchestral overdubs, but somehow we made it happen. On "Jeepster," for example, I used four cellos and a bassoon to play those descending bass lines. It was a good period of my life to learn and devise new tricks to make some very simple rock tracks into stunning productions.

In addition to your conventional orchestration skills, I've always loved the way that you've used sonic textures as "hooks."

I acknowledge that I am a sound addict. I love enhancing and changing the sound of an ordinary instrument

▼

"I'm a firm believer in keeping everything the artist puts on tape and then editing the best performances together on a composite track."

and coming up with something that has never been heard before. Robert Fripp's guitar tone on Bowie's "Heroes" was filtered, ring modulated, and otherwise mangled through Eno's briefcase synth. Although the guitar plays a strong theme, the sound is the thing that is mesmerizing. Rock music is mainly about energy and sound.

Also, a conventional melodic part can really come alive when its timbre is rendered "unconventional." The riff at the beginning of "Ashes to Ashes" on Bowie's *Scary Monsters* album, for example, started out as a simple piano motif. But the part gained a wonderfully eerie presence after it was strangulated through an Eventide Instant

PAUL CARTLEDGE

Visconti recording *The Other Side of Life* with the Moody Blues' Patrick Moraz (left) and Justin Hayward (right) in his own Good Earth Studios in 1986.

Flanger that only had one side of the stereo output working. The first choice for the motif was a Fender Rhodes, but we couldn't get one right away. I'm glad we didn't.

But what inspired fit of madness caused you to process a snare drum through a Harmonizer on Bowie's *Low* album?

I owned one of the first Eventide Digital Delays, so I received a press release about their Harmonizer that claimed it could change pitch without changing speed. This was science fiction to me, and no matter how much it cost, I had to have one. Now, as the manual suggested auditioning instruments and voices through the Harmonizer, I stayed up all night processing every single track on a multitrack tape. When I put a snare drum through it—while decreasing the pitch and adding feedback—I heard the heaviest snare of my life. It was truly magic, and I couldn't wait to try this thing out on a commercial recording.

About that time, Bowie asked whether I'd mind making an album with Brian Eno in France, and we commenced to make *Low*. I unveiled my secret weapon, patching the snare mics directly into the Harmonizer and recording the effect on track 24. When drummer Dennis Davis heard the sound, he begged to have it routed into his headphones. We soon discovered that the rate of the Harmonizer's drop off was controlled by an envelope at its input. So now that Dennis could hear the effect as he played, he was able to control the sound by how hard he hit his snare. This is why hardly anyone has duplicated that snare sound—we didn't do it in the mix, we did it live!

It seems obvious that experimentation and "happy accidents" are major components of your production style.

Yes, accidents are always waiting to happen. I'm delighted when I push up the wrong fader and discover the guitar amp blasting through a vocal mic from 30 feet away. If it sounds good, I record it and don't ask questions.

Another happy accident is when I carefully rehearse background singers or an instrumentalist to end a section on a particular note or chord. However, when the time comes, they forget and hit something so far out, I never would have thought of it. If it sounds incredible—even if it wasn't planned—I just have to go with it. I love it when stuff like that happens.

I know that you're extremely proud of your home studio, but isn't it difficult to work at home when you've tracked in the best studios the world has to offer?

Believe me, my digital-editing facilities at home are far better than in any studio I've ever worked in. I can bring tracks home from anywhere I work and do some scary, tricky stuff with my Macintosh and MIDI equipment. I can edit drum tracks, "tune up" vocals and guitar solos, and record overdubs at home and simply transfer the results back to whatever format I was using in the large studio. Then, I can mix in the grandeur of an SSL or Neve room.

And there's another benefit to my home studio: I can take on projects that tickle my fancy but lack heavy financial backing. This situation is directly responsible for my being able to work with Alex Forbes, a very cool songwriter who wrote "Don't Rush Me" for Taylor Dayne. Alex has another one of those "quirky" voices—and she writes brilliant lyrics—so I casually entered a writing/producing relationship with her.

We now have a completed album that contains eight mutually penned songs. And although the project was recorded at home, it has Richie Morales on drums and Noel Redding and my son, Morgan, on bass. I am very proud of this album. I've often felt that I started my music career as a songwriter but got waylaid into being a producer for 28 years. Now I think I'm on the right track again, thanks to my home studio and Alex.

Now I *do* want to make clear that although I love my little home studio, I still enjoy having a million-dollar console wrapped around me in a world-class facility. My home studio is just an alternative and serves different purposes.

Obviously, you've done your share of engineering, and you continue to engineer at home. Is it difficult to turn the console over to someone

Visionary Abode

Producer Tony Visconti has been extremely adept at using "Class A" studios to record and mix while bringing tracks home on ADAT slave reels for overdubs, sequencing, and digital editing. He didn't want to list *all* of his personal studio gear, but here's a little insight into what Visconti uses to wring the best out of both audio worlds.

Mixing Console	Soundtracs Solo MIDI (32 x 8 x 2)
Recording Media	Alesis ADAT (3) with BRC controller, Panasonic SV-3700 DAT recorder
Monitor Speakers	Alesis Monitor One
Microphones	Audio-Technica AT4033, Neumann U87 (2)
Keyboards/Sound Modules	Akai S900 and S1000, E-mu Proteus/2, Korg Wavestation SR, Roland Jupiter-8, Roland JV-1080, Roland MKS-20
Signal Processors	Various effects units from Alesis, Lexicon, and Yamaha; twelve different compressors; Manley Direct Tube Interface; RSP Saturator
Computers/Software	Mac Centris 650, Digidesign Sound Designer II, Emagic *Logic Audio*

THE WORLD'S BEST SELLING MICROPHONE

NOW COMES WITH THIS HANDY ACCESSORY

GET A $10 REBATE ON THE SM58 MICROPHONE

FROM SEPTEMBER 1ST THROUGH OCTOBER 31ST 1995

The Shure SM58 is the standard by which all other microphones are measured. And for a limited time, we've added something that will make the SM58 even better: a $10 rebate.

But hurry. The SM58 is designed to perform for years, but this $10 rebate offer ends soon.

SHURE®

SHURE'S "ACCESSORY" REBATES:

To receive your rebate send: 1. A copy of your dated sales receipt (non-returnable) indicating model number and name of store where mic(s) were purchased, 2. The actual silver model number label (no substitutions) from the end of each outer carton, and 3. This completed coupon to: Shure "ACCESSORY" Rebate Offer, 222 Hartrey Ave., Evanston, IL 60202-3696

Name _____

Address _____

City _____ *State* _____ *Zip* _____

Phone _____ / _____ *Qty. purchased:* _____

$10 rebate applies to Models SM58-LC, SM58-CN and SM58S only. Wireless version not included. Rebates are limited to a maximum of four per customer, household, family, or organization. Offer valid only on consumer purchases made at retail between September 1, 1995 and October 31, 1995. Rebate claim forms must be postmarked no later than November 31, 1995. This is a consumer (end user) offer only. Shure microphone distributors, retailers, their employees or their families are not eligible for rebates, nor may Shure microphone distributors, retailers, their employees or their families claim rebates on behalf of consumers (end users). Rebate requests not including proper documentation (official coupon, actual silver model number label (no substitutions) from end of carton, and dated sales receipt with retailer's name) will be ineligible. Shure is not responsible for late, lost, or misdirected mail; insured or certified mail is recommended. Offer good only on purchases made in the U.S.A. and open to U.S. residents only. Void where taxed or prohibited by law. Allow 6 to 8 weeks for delivery of rebate check. Coupon face value 1/4¢. Rebate eligibility as determined by Shure is final.

else when you're producing a session in a large studio?

No matter where I work, I do most of my own engineering. I've been working like this for 25 years, so it isn't that hard for me to engineer and produce simultaneously. However, making records requires teamwork, so I'll often work with another engineer or a talented assistant. If the engineer is really hot, I will rarely interfere. I'm always open to learning about different recording techniques and new microphones, so I turn the experience of working with an engineer into a personal seminar. But if I'm looking for a very specific sound and I know exactly how I want to get it, I have no problem giving the orders. Too much is at stake to allow everybody to put in their two cents.

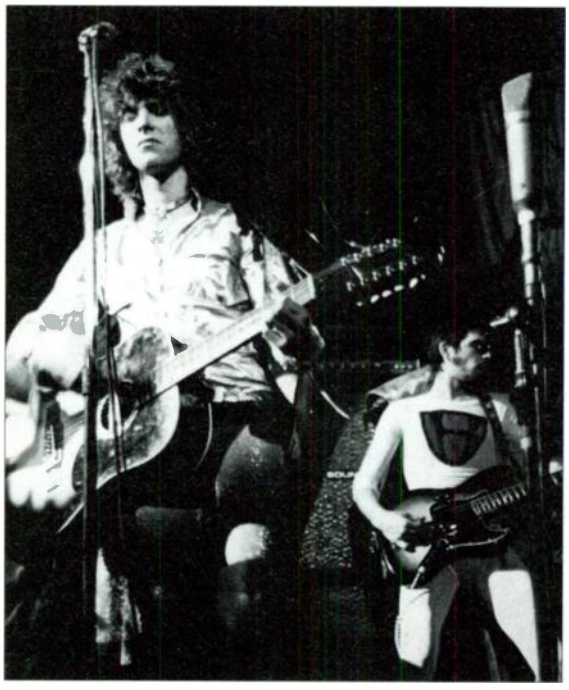
Do you feel a producer must have engineering chops or just an intuitive ear for music?

Without a doubt, you need some recording chops. After all, this *is* a profession. Making a record has become a highly technical procedure that requires a vast knowledge of music and technology. You can never know enough. Every month, I read all the journals pertaining to my profession and have regular talks with my colleagues about new equipment and how it works. As a musician, I make it a regular practice to jam with my mates and listen to as much new music as I can stand. Knowledge is power.

What are some of the major differences between being a producer today and when you started your career in the late 1960s?

There are so many more ways of recording music now. I was fortunate enough to start out when 4-track recorders were pretty much all that was available. My learning curve was slow and steady. I pity a producer starting out today!

Cut and paste wasn't even *conceived* in the days of T. Rex, and two machines couldn't be locked up until the early 1980s. We did everything on the fly. We sped up and slowed down tapes, we put buckets on our heads, and we swung mics around the studio like lassos to emulate the Doppler effect. We had very few tools capable of making fantastic sounds, so we taxed our imaginations to reproduce the noises we heard in our heads. And my generation was living in the shadow of the Beatles, who made fantastic, surreal recordings with even less equipment than we had!



RAY STEVENSEN

Real producers don't wear tights? Visconti (right, on bass) and David Bowie in the aptly named band Hype, in 1969.

However, it is still the producer's job to record music, and the responsibilities have remained the same throughout the years. You work within the confines of a budget and strive to cut a record that makes such a shocking contribution to the culture that it is considered a classic. So, in a sense, nothing has really changed except the toys.

It was Tony Visconti's striking and evocative productions for T. Rex and David Bowie that inspired **EM** *Editor* **Michael Molenda** *to start mucking about in recording studios.*

Illustration by David Peters; Photos by Jacki Sallow.

Wright at Home

Gary Wright weaves his musical dreams out in the garage.

By Greg Pedersen

BACK IN THE DAYS OF SHAG HAIRCUTS, rock stars and label presidents alike seemed more interested in clutching bottles of Dom Perignon than maintaining firm grips on their company's purse strings and recording budgets. The whole record industry was like one big party, and the festivities didn't stop at the studio door. Producers and engineers were often given carte blanche to do whatever it took to create a masterpiece. As a result, sessions dragged on for months, ensuring that the studio owner made as much money from an album as the artist who recorded it.

Although many of his peers reveled in the era's free-spending attitude, keyboardist Gary Wright realized the easy ride wouldn't last. So after conquering the pop world with his smash hit "Dream Weaver" in 1976, Wright spent part of his career jackpot transforming the garage of his Palos Verdes, California, home into High Wave Studios.

"I figured that I was going to be making records for a long time," says Wright. "So it seemed rather senseless to support a bunch of recording studios and pay out thousands of dollars per album when I could take that money and invest it into my own home. It cost approximately $40,000 to build High Wave, and the studio has paid for itself more than three times already."

Wright at Home

Wright's longevity is certainly the envy of more than a few of the hitmakers from the 1970s. Whether the public considered him way cool or totally uncool, Wright never stopped making records. And now, the catchy world beat-flavored pop of his latest album, *First Signs of Life,* is all over Adult Contemporary airwaves. Although Wright did a little globetrotting to breathe some fire into *First Signs of Life,* the project ended up right back in his own garage.

CARNAVAL CRASHING

Current recording technology makes it child's play to go into the field and hunt for authentic or unique sounds and then bring the tracks back to the comfort of one's personal studio. Therefore it made perfect sense for Wright to seek out "real" drummers for *First Signs of Life* rather than sequence a batch of ethnic drum samples. Next stop: Rio de Janeiro.

"I went down there at Carnaval time, which is when they have these amazing parades with all these percussionists marching in them," explains Wright. "I got the best drummers into a local recording studio and had them play along with some tempos I tapped out. I also played a couple of songs on acoustic guitar with the percussionists providing the rhythm tracks."

When the tapes arrived back at High Wave, Wright sampled some of the performances in 8-bar sections to use as rhythm loops. Typically, three or four loops were pasted together for each song, and any slight rhythmic imperfections were left intact to maintain a human feel.

"I didn't want the loops in mathematically perfect time, because they wouldn't sound like real people playing," he says. "It's only natural that there's a little fluctuation in the 8-bar loops. As long as the pattern comes back on the 'one,' everything should sound okay."

Most of the rhythm loops were designed using Wright's favorite sampler, the E-mu EIIIx. However, a couple of the tracks on *First Signs of Life* did not use loops from the Carnaval sessions. For these songs, Wright took the more conventional home-studio approach of MIDI sequencing.

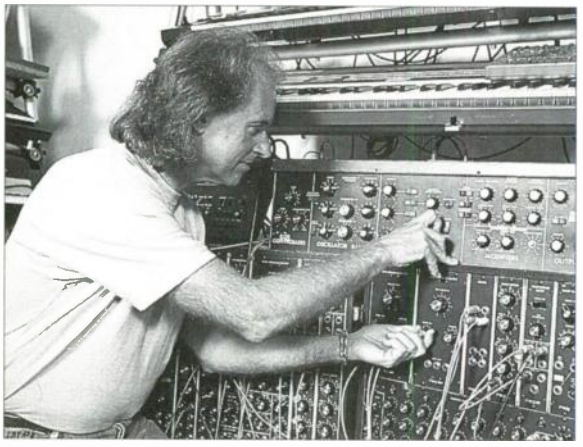
RETRO REVERIES

Although the equipment list at High Wave has seen a lot of changes through the years, Wright is not afraid to stick with what works for him. He has not upgraded to modular digital multitracks or hard-disk recorders because his trusty 2-inch, analog 24-track deck is doing just fine.

"I simply prefer analog to digital," Wright declares. "The only way I can explain it is with an analogy of rubbing your fingers over velvet or silk. You can feel a textural difference between the two fabrics. Well, I hear a textural difference between analog and digital recordings. There's a certain amount of tape compression and harmonic distortion that occurs in the analog realm that is very pleasing to my ear. Digital is great for its extremely low noise floor, but I

haven't found a hard-disk recording system that is so technologically advanced that it sounds as warm as analog."

Wright's taste in keyboards is similarly backdated. He picked up his first synthesizer, a Moog modular system, while playing with the band Spooky Tooth in the early 1970s and readily admits that his current synth collection runs three-to-one in favor of analog models. However, he is not starry eyed about the past and is quick to point out that analog synthesizers had their share of frustrating problems.



JACKI SALLOW

Next time you want to complain about nested parameter menus, just remember all the patch cords and knobs Wright has to plug and unplug to program his Moog.

"Those old synthesizers were a tuning nightmare," recalls Wright. "You'd have to tune each oscillator, and then they would still go out of tune. And the Moog modular systems were huge! They would practically take over the whole studio. If I wanted to put a synthesizer solo on a song, we'd have to schlep in this mammoth synthesizer and then bring in somebody who knew how to tweak all the oscillators. Believe me, those Moogs were a roadie's nightmare."

Sequencing was also very tricky throughout the "good old days." In the early 1980s, technology hadn't quite advanced at the same rate as the creative ideals of keyboardists such as Wright.

"I try to forget about the early sequencers," he says. "They were always dumping memory, which was a huge problem. I had a Linn, and I'd spend an entire day programming the thing only to have it say, 'memory erased'! There went my song. In the real early days, I'd often do *faux* sequencing by recording sounds from the Moog directly onto tape. Of course, we didn't have synchronization devices like they do now, so the drummer would have to play along to whatever the Moog was doing. The modulating synth became the time reference."

HOME STYLE

Wright's home studio has witnessed the technological advances of analog recording, MIDI, and digital audio. However, the foundation of Wright's

The Wright Stuff

Mixing Console	Amek Matchless 38 x 24 x 2
Recording Media	Ampex MM1200 analog 24-track, JVC CR6500 VCR, Panasonic SV-3700 DAT recorder
Monitor Speakers	Hemisphere, Tannoy FSMU
Microphones	AKG C 452; Groove Tubes Model 1; Microtec Gefell; Shure SM7, SM57
Keyboards/Sound Modules	Alesis SR-16; E-mu EIIIx, Emax; Korg 05R/W; Moog Memorymoog, Minimoog, Micromoog, System 15; Roland D-50, JX-8P, MKS-30, MKS-80, R-8; Yamaha DX100, TR812, TX81Z
Signal Processors	dbx 165A; dbx 162; Lexicon 200, LXP-1, LXP-5, MRC controller; UREI 1176LN, LA-12; Yamaha REV7
Computers/Software	Macintosh IIci, Digidesign SampleCell II, MOTU *Performer*
Power Amp	Boulder 250AE

Wright at Home

creative world—the studio itself—has withstood the ravages of time because it was built to last.

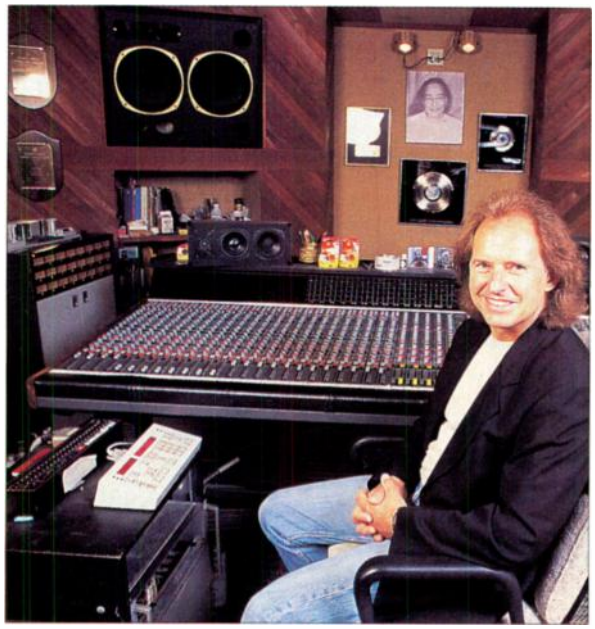
"I made sure I did everything by the book," says Wright. "I had an engineer come out and design the room so there were no parallel walls that could cause standing waves and other acoustical problems.

"Also, I had the studio completely soundproofed by doing the room-within-a-room design," he continues. "This type of construction allows for dead air space between the double walls, and it really seals out environmental noises and prevents studio sound from escaping. Soundproofing was essential to my creative comfort. You see, a lot of people who record music in their homes have to do everything real quiet and monitor through headphones, because you can hear the session all over the house. I'm not saying that you have to soundproof your studio to do good work, but it sure changes the creative experience when you have to watch the noise level."

Although Wright employs samplers, sequencers, and drum machines to record his rhythm tracks, he is not limited to electronic grooves. In fact, when he first established his studio, he used the only drum machines available—humans.

"Like most home studios, High Wave wasn't designed for full-blown acoustic drum sessions," admits Wright. "It's an electronic music studio. But whenever I've needed to record live drums, I've been able to improvise quite successfully. Usually, I'd just take a room in my house and set up the microphones! It's really all a question of miking. The room doesn't have to be acoustically perfect. All you need is a reasonably big, live-sounding space and a few good mics to get some great sounds."

However, Wright does not improvise or cut corners on his monitoring system. He feels that without proper monitors, you can't produce high-quality work. According to Wright, if you can't



JACKI SALLOW

Gary Wright's well-equipped home studio lets him tear it up like a one-man garage band.

critically assess your sonic options, no amount of technological wonders will improve your recordings.

"If you have good near-field monitors, you don't have to spend money trying to get your control room sonically perfect," he explains, "because you have an accurate reference right up close to your ears. I've used all types of monitors, and I've found that using near-fields really improves the accuracy of my mixes. The only area where they fall down a bit is in the low end, so I recommend using big monitors to make sure you aren't adding too much bottom. However, I swear by these near-fields called Hemispheres; they were designed by an engineer, Jay Lewis, who worked with me on the *Dream Weaver* album. Unfortunately, you can't get them anymore."

DREAM WORLDS

As a veteran of both home and large studio sessions—and years of recording experience as a band member and solo artist—Wright contends that working at home best "fits his personality."

"Now, with the tools I have at home, it's possible for me to do an entire album all by myself," he says. "That's great, but there *is* a trade-off. I sometimes miss the excitement of working with other musicians, because working by yourself or with one other person can be a little bit boring."

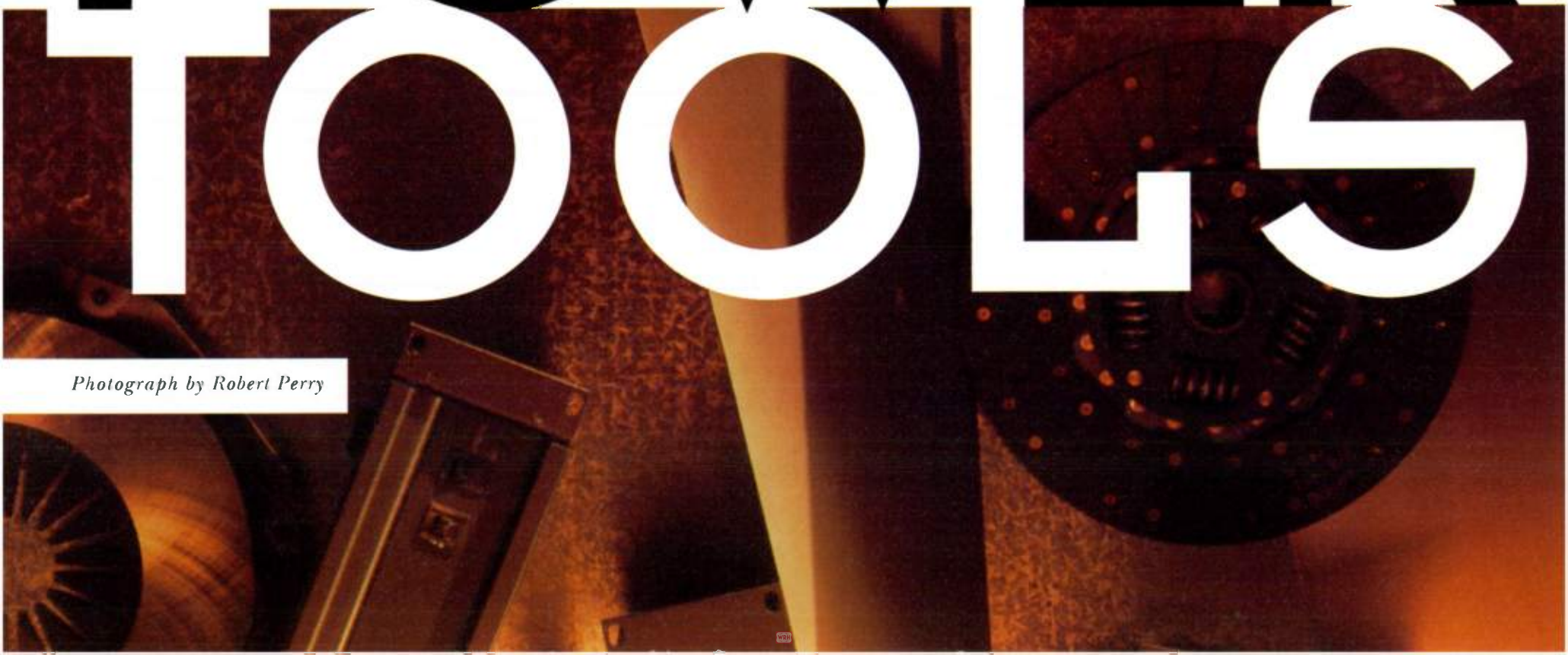
And although Wright's well-equipped home studio will, by design and intention, never rival big techno palaces such as the Record Plant, he doesn't view limitations as a bad thing.

"Back when you didn't have everything at the touch of your fingertips," maintains Wright, "there wasn't a tendency to get lazy. When you had very little to work with, you tended to get a lot more creative with what you had."

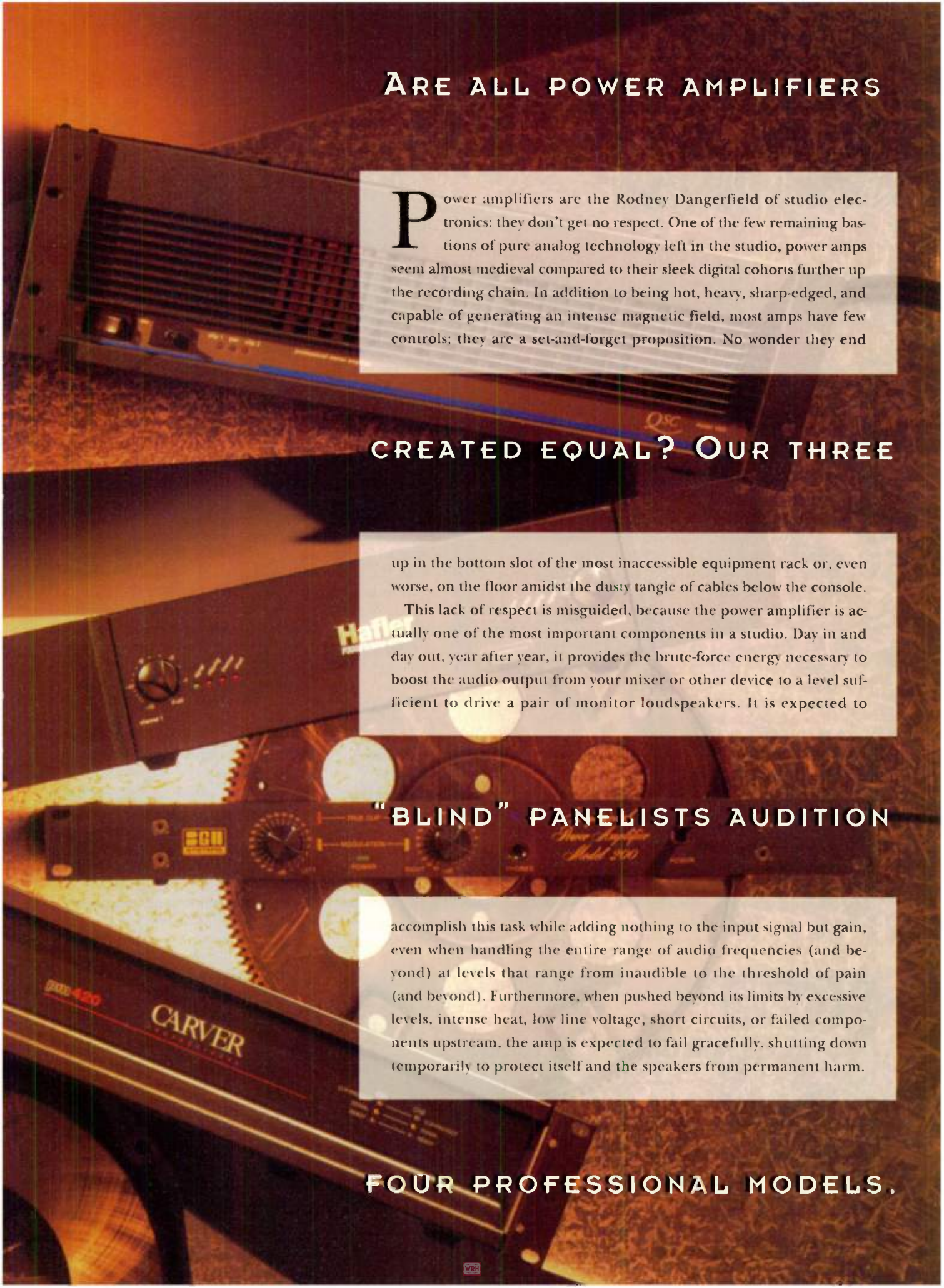
Greg Pedersen *is a freelance writer living in the San Francisco Bay Area.*

By Lawrence E. Ullman

POWER TOOLS



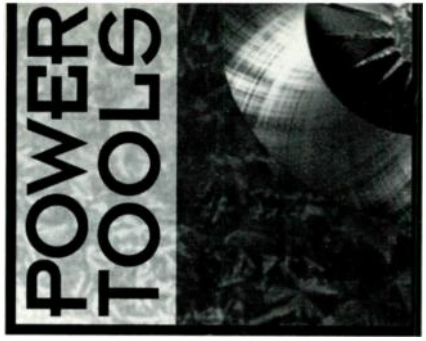
Photograph by Robert Perry



ARE ALL POWER AMPLIFIERS CREATED EQUAL? OUR THREE "BLIND" PANELISTS AUDITION FOUR PROFESSIONAL MODELS.

Power amplifiers are the Rodney Dangerfield of studio electronics: they don't get no respect. One of the few remaining bastions of pure analog technology left in the studio, power amps seem almost medieval compared to their sleek digital cohorts further up the recording chain. In addition to being hot, heavy, sharp-edged, and capable of generating an intense magnetic field, most amps have few controls; they are a set-and-forget proposition. No wonder they end up in the bottom slot of the most inaccessible equipment rack or, even worse, on the floor amidst the dusty tangle of cables below the console.

This lack of respect is misguided, because the power amplifier is actually one of the most important components in a studio. Day in and day out, year after year, it provides the brute-force energy necessary to boost the audio output from your mixer or other device to a level sufficient to drive a pair of monitor loudspeakers. It is expected to accomplish this task while adding nothing to the input signal but gain, even when handling the entire range of audio frequencies (and beyond) at levels that range from inaudible to the threshold of pain (and beyond). Furthermore, when pushed beyond its limits by excessive levels, intense heat, low line voltage, short circuits, or failed components upstream, the amp is expected to fail gracefully, shutting down temporarily to protect itself and the speakers from permanent harm.



Yet choosing a power amplifier for studio use is far from a straightforward process. Marketing hype and obscure jargon abound, "specmanship" runs rampant, and comparative listening tests are essentially impossible to conduct in any retail environment. To help guide you through this morass, I gathered together four representative, mid-range, professional power amplifiers: a BGW Model 200 (100 watts/ch., $998); Carver pm420 (135 watts/ch., $695); Hafler P3000 (150 watts/ch., $799); and QSC Series One Model 1200 (100 watts/ch., $698).

We're going to get down and dirty with these amps to see what makes them tick. I'll explain and compare their features and specifications, concentrating on those that are meaningful and useful in the real world. Then, I'll pop the lid on Pandora's box and invite a few unsuspecting golden-eared friends over for a "by-the-book" listening test. By the time we're done, we should be able to verify or refute the oft made claim that "these things all sound the same, anyway." But before we start the tour, we need to have a little discussion about...

THE ROYAL FLUSH

Despite the great demands placed on any power amp used for studio applications, many people seem quite willing to skimp on this critical component. It's not unusual to see wheezing old consumer receivers and integrated amplifiers pressed into service in otherwise well-equipped studios. (The home studio profiled in the August 1995 "Creative Space" feature is a recent example from these very pages.)

Don't get me wrong: consumer-grade amplifiers are fine if you are a hobbyist starting out on a tight budget; your limited funds are better spent elsewhere. However, if you're serious enough about your recording projects to have invested in a modular digital multitrack, quality mixer, condenser mics, signal processors, and so on, it's time to consider moving up to a more capable professional power amplifier.

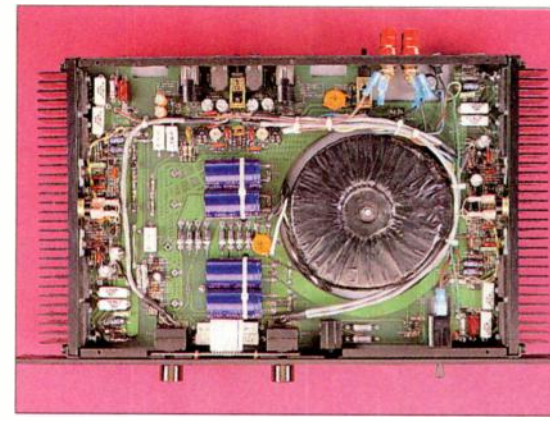


FIG. 1: No guts, no glory. Heavy-duty power supplies are important factors in most professional power amplifiers. The BGW 200's power supply is served by a huge, toroidal (doughnut-shaped) transformer; four large power-supply capacitors lie to its left.

To understand the differences between professional and consumer amplifiers, you need to know a little bit about amp construction. But let's get one thing straight right off the bat: this has little to do with the various circuit designs or "topologies" used. For the record, one amp in the test group—the Hafler P3000—produces power with MOSFET transistors and trans•*nova* (Transconductance Nodal Voltage Amplifier) technology; the others use standard Class AB construction and bipolar transistors.

Although MOSFETs have certain advantages over bipolar transistors—they behave more gracefully when they're overheated, shutting down gradually instead of failing dramatically—designs using them are not inherently better or worse than others, just different. If the minutiae of amplifier design make the propeller on your beanie spin, **EM** has published two articles you should check out: "The Power and the Glory" (August 1993) and "Basic Studio Series, Part 2: Power Amplifiers" (December 1989).

The most important difference between pro and consumer amps is the relative capacities of their power supplies (see **Fig. 1**). Forming the heart of any amp, the power supply includes a large *transformer* that feeds several electronic devices called *capacitors*. Functioning like a battery (albeit one with a *very* short shelf life), the capacitors store an electrical charge. The transformer's job is to keep them fully topped off at all times. In effect, the power supply provides a reservoir of voltage. The *output transistors* then tap that reservoir, drawing current from the power supply in direct proportion to the input signal.

The design of an amp's power supply has a direct bearing on how well it handles repeated transients. "Consumer units are notorious for using the minimum power transformer they can get away with and trying to make up for it with capacitive storage," says Jeff Phillips, marketing manager at BGW. "That's okay for the first transient, but once you've drained the capacitors for that peak and you go back to do the next one, the capacitors are not fully recharged yet. You end up with a lack of punch; the amp just wimps out. If you have a real pro amplifier, its going to have a big enough transformer to recharge the capacitors between peaks.

"Let's put it another way," Phillips continues. "The difference between a consumer amp and a pro amp is like the difference between a residential

Looking more like a signal processor than a power amp, BGW's Model 200 packs 100W/channel into a single rackspace.

CDs IN 3 WEEKS!

300 CDs plus 300 Cassettes for only $2,390

with two-color inserts & chrome tape

Includes FREE design!

Here's what you get from the leader in the independent music industry.

FREE deluxe graphic design
Done by some of the finest artists in the business. Your new release will jump off the shelf!

Express Manufacturing
In a hurry? Express CD packages are available in just 3 weeks, cassettes in 7 days. Includes design!

Proof Positive™ Reference CD
We FedEx our exclusive test CD for you to approve so you'll know exactly what your finished product will sound like.

Two-Day Shipping
Whether you live in Boston or Los Angeles, two-day shipping at low ground prices is our policy.

Money Back Guarantee
There's no fine print. You must be completely satisfied with you proofs and tests or we'll cheerfully refund your money.

"I chose to return to Disc Makers for my second project because your staff took the initiative and really put their experience to work for me. The feedback I have received on everything – from the graphic design and insert printing to the high-quality audio and promotional posters – has been incredible!"

–Joe Restivo
Astoria, NY

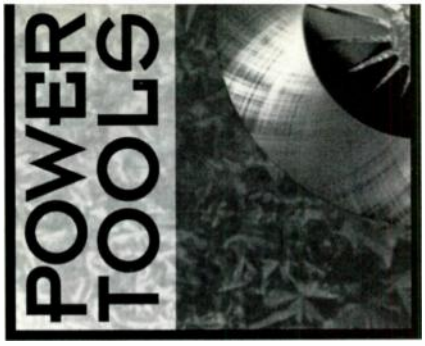
Call today for your FREE, 1996 full color catalog:

1-800-468-9353

24 HOURS TOLL FREE
Outside USA call 609-663-9030; FAX 609-661-3458

DISC MAKERS

AMERICA'S #1 MANUFACTURER FOR THE INDEPENDENT MUSIC INDUSTRY



and a commercial toilet. A one-shot transient is like flushing the toilet in your home; no problem, it's going to cycle just fine. Just don't give it another 'transient' for a while! The toilets at Chicago's O'Hare airport, on the other hand, can be flushed rapid-fire with no wait in between, as they have a much higher 'duty cycle.' The transformer is the size of the pipe coming into the toilet; a big 1.5-inch industrial pipe, for example, is going to go 'whoosh' and fill up the tank in a hurry. The tank is the storage capacitors in your power supply. And having stored energy right there to handle transient after transient is the key to having a really good monitor amp."

Another key difference is the amp's ability to deal with heat. Amplifiers are inherently inefficient, converting as much as half the AC power they consume into heat. Because the output transistors can be destroyed by excessive temperatures, some form of cooling must be provided. All the amps in this test group rely on silent convection cooling instead of noisy fans, although a "whisper" fan can be easily retrofitted to the Carver and QSC if desired.

The cabinets of all the amps but the QSC bristle with *heat-sink* fins; these provide plenty of surface area and a large *thermal mass* to conduct heat away from the power transistors. Rather than exposed fins, the QSC uses its heavy, louvered front panel as a heat sink, supplemented by smaller heat sinks inside the cabinet. By comparison, consumer amps—especially receivers—have much less thermal mass and must make due with relatively tiny heat sinks mounted inside the cabinet.

BGW's Phillips also points out that most pro amps use more output transistors than similarly powered consumer models, which lessens the thermal load each transistor must endure. "What really ages an amp is thermal cycles on the output transistors," he says. "We build our amps to last ten years or longer. For example, our Model 200, which is nominally 100 watts per channel into 8 ohms, has enough output transistors to provide 1,600 watts of heat dissipation. Spreading the work over more transistors is more expensive, but it keeps the heat rise on any one transistor down, and this allows them to last a long time."

The bottom line is that pro amps are designed to survive heavy-duty, 'round-the-clock use. As long as you don't physically abuse the amp (and perhaps even if you do), it will stand up to the rigors of studio operation for years. **EM** Technical Editor Scott Wilkinson and I can personally attest to the longevity of the QSC Model 1200: I bought one over fourteen years ago and sold it to Scott several years later. It's still going strong in his studio.

Before we go any further, let me point out that my definition of the term "consumer amp" does *not* include high-performance *separate* power amplifiers designed for high-end audio and home-theater applications. Some of these are endowed with bigger power supplies, more heat-sink area, and more output transistors than any pro amp. However, most "audiophile" amps lack many of the other features typically found on pro models, features that may or may not make a pro amp.

THE RIGHT TOOL FOR THE JOB

Because professional amps are designed for sound-reinforcement, P.A., touring, and musical-instrument duties, as well as studio use, they are expected to offer certain features that further differentiate them from their consumer counterparts. Of course, whether you need these features depends on your intended application.

One of the most obvious distinguishing characteristics of a pro amp is a roadworthy, rack-mountable cabinet. The cabinets in our test group are diverse in design and provide a good cross-section of those found on most pro amps. The Carver pm420 is solidly wrapped in a traditional sound reinforcement-style cabinet, with the front panel recessed to prevent damage. The 2U cabinet is the deepest of the group; rear rack ears are provided and should be used if at all possible. The one-piece top cover wraps all the way around the sides, cleverly protecting the heat sinks, which are revealed through large openings. A small gap at the top of the front panel is hidden from view by a lip that projects down from the top cover. This gap allows air to flow through the amp and is especially useful if the optional fan is installed.

The Hafler P3000 is the iconoclast of the group. Although very strong and roadworthy, its cabinet seems to have been designed with more sedentary studio use in mind. In particular, the front panel projects slightly forward of the rack ears, which places the panel and its controls slightly closer to harm's way. Like the Carver, the P3000 occupies two rackspaces, but it is much more shallow. In fact, the cabinet appears tiny compared to the other amps. Much of the unit's dimension comes from massive cast-aluminum heat sinks that form not only the sides of the unit but the rack ears and several inches of the front panel, as well (see **Fig. 2**).

The QSC requires three rackspaces but is almost as shallow as the Hafler. As previously mentioned, its cabinet is the only one with no external heat

Power Amplifier Features

	BGW	Carver	Hafler	QSC
Short Circuit Protection	yes	yes	yes	yes
Thermal Overload Protection	yes	yes	yes	yes
DC Offset Protection	yes	yes	no	yes
AC Circuit Breaker	yes	yes	no	yes
Other	n/a	clipping eliminator	line fuse	RF protection
Ground-Lift Switch	yes	no	yes	no
Indicator Lights	P, S, C	P, S, C	P, S, C, T, S	P, C
Dimensions (H x W x D, in inches)	1.75 (1U) x 19 x 11	3.5 (2U) x 19 x 13.25	3.5 (2U) x 19 x 10	5.25 (3U) x 19 x 9.6
Weight (lbs.)	14	24	26	24
Price	$999	$695	$799	$698

Legend: P=Power, S=Signal, C=Clip, T=Thermal, S=Short

There are regions of experience that exist only in the mind of creative genius. Only a Lexicon can unleash this potential.

PERFORMANCE SERIES

So Many Ways to achieve that Lexicon Sound.

The ***Lexicon*** *tradition of sound quality is so highly respected that over 80% of recordings made in the world today utilize a Lexicon reverberation or effects processor. And while a Lexicon is an integral component in practically every World Class audio production environment, there's also a Lexicon processor for the equally demanding needs of project studios — as well as performing musicians. Every one offers the superlative Lexicon sound: from the powerful* ***PCM-80*** *— with its 24-bit digital bus and dual-DSP architecture — to the affordable* ***Alex*** *&* ***Reflex*** *reverberators. Then there's* ***Vortex****, with its unique Audio Morphing between effects.* ***JamMan****, the delay looper. The LXP Series, including the* ***LXP-15 II*** *with its new software, and the compact* ***LXP-5****. And the* ***MRC MIDI Remote Controller*** *to tie them all together. Now, aren't you glad there are so many affordable ways to get that Lexicon sound?*

Don't you deserve the Lexicon Sound too?

lexicon

Heard In All The Right Places

Contact! 100 Beaver Street, Waltham, MA 02154 Tel: 617/736-0300 Fax: 617/891-0340 E-Mail: 71333.434@compuserve.com

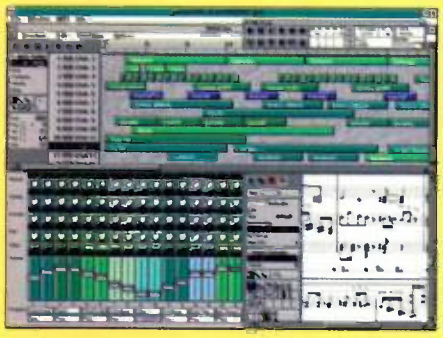
A Harman International Company

Music Software's Finest

MicroLogic

Making music has never been so easy! Affordable with a professional touch.

MicroLogic for Windows™ and Macintosh™ introduces you to the world of computer aided composition with a package boasting a range of powerful features you won't find elsewhere in this price range. There are three professional editors to help you achieve those perfect final results in your music. The unbeaten resolution of 960 ppqn gives you the best timing available today and the integrated General MIDI mixer offers you the perfect balancing for all GM sounds



and effects. User definable color coding for all tracks and a powerful interactive notation section provide maximum overview and easy handling. Should you one day desire more functionality and power, you can simply upgrade to Logic.

Logic

The most powerful and modern music software available for MS Windows™.

Logic for Windows is clearly the leader in integrated MIDI Recording and Notation systems on the PC platform. With an unlimited number of tracks, a large choice of quantization algorithms, our unique Environment for creating virtual MIDI studios and mixer consoles, up to 90 definable screensets, interactively linked windows, a seemlessly integrated high end scoring section, freely definable key commands for virtually every function, Logic is the

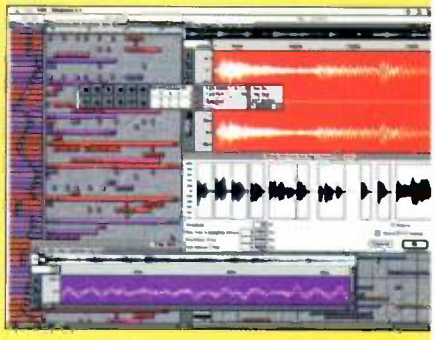
perfect tool for producing music in professional or home studio environments. Logic for Windows (also available for Macintosh) will offer you entirely new possibilities and its flexibility will amaze you again and again.

Also available as 100% native code version for PowerPC!

Audio Module

The add-on integrated module that turns Logic into a total digital studio!

Install the Logic Audio Module with Logic on your Mac/Power PC, and all doors to the exciting world of digital audio recording will swing wide open. You can use the Apple Sound Manager to record* and play up to 12 tracks or use any of the audio hardware options from Digidesign (4-48 tracks**). Arrange MIDI and audio tracks on one surface, edit the pitch and length of a digital audio track in one process, adapt the groove of your MIDI tracks to the groove of an audio track, turn monophonic digita



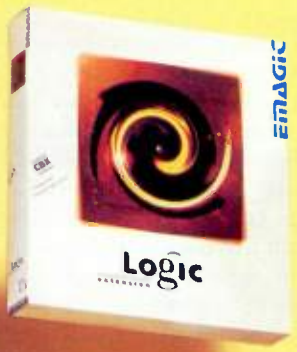
recordings into printed score and enjoy what many users say is the best music program ever made.

The Logic Windows Audio Module is scheduled for Q1/96

* *16 bit Audio-in Mac only*

** *16-48 tracks using TDM Extension*

Jewels. The Logic System.



TDM Extension

The Logic TDM Extension will make your Pro Tools III a total state of the art Digital Audio Workstation!

In combination with Digidesign's Pro Tools III system and Logic Audio, the TDM Extension allows for 16 - 48 digital audio tracks. Thanks to a special Audio Object in Logic's Environment you can create virtually any Pro Tools mixing desk that you may need. All Plug-Ins (digital effects) can be fully automated including Aux Sends and Master Outputs. Plug-Ins can also be inserted and deleted during playback and every possible TDM configuration can be saved

with your songs. Depending on your hardware, you can use as many Plug-Ins (inserts) per track and for the final mix/sum as you may need. Each of your TDM configurations can be intuitively copied from song to song. Digital audio recording luxury? This way you can...

Logic Audio

The Modern Masterpiece. Integrated MIDI and Digital Audio Recording for those who require only the best.

Logic Audio is our flagship product in the Logic Series. This package integrates the complete Logic program with the Audio Module. You get up to 12 tracks using Apple's Soundmanager (AV), 4 - 48 tracks** with one of Digidesign's hardware options, or 4 - 8 tracks with the Yamaha CBX D3/D5 (add CBX Extension). Included at no extra charge is the legendary Digital Factory, our collection of power utilities for creative DSP editing of your audio material. The Digital Factory™ includes

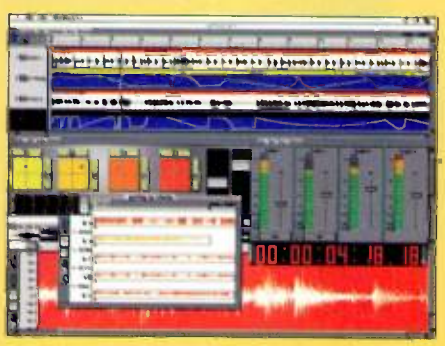
the Time Machine™, the Quantize Engine™, the Silencer™, the Groove Machine™ , the Audio Energizer™, the Audio to Score Streamer™, Audio to MIDI Groove Templates™ and more. You owe it to yourself to seriously check out this program before you consider buying anything else.

** *16-48 tracks using TDM Extension*

CBX Extension

Affordable digital audio tracks using the powerful Yamaha CBX D3 or D5 system.

Control the innovative CBX D3 and D5 Digital Recording Processors with Logic Audio using the CBX Extension. With each CBX unit you add an extra 4 digital audio tracks, 4 outputs and 2 inputs for up to 8 CBX tracks. Logic Audio is the only program that lets you use all Mac/PPC (AV), Digidesign and CBX tracks simultaneously including all options of the powerful Digital Factory.



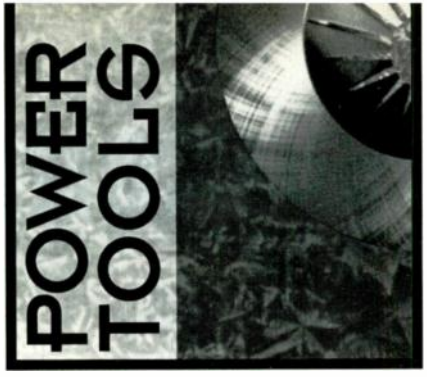
Ask your dealer for a demonstration today!

EMAGIC

We make computers groove

EMAGIC Inc.
Tel. 916-477 1051
Fax:916-477 1052

Digidesign Development partners booth #718



sinks; the recessed front panel has louvers that serve this function. The BGW 200 is only one rackspace high but is quite deep. It has a thick aluminum faceplate and looks more like a high-quality signal processor than a power amp. It also has knobs and a power switch that project well beyond the face plate; care should be taken to protect these from damage.

The knobs on the BGW's front panel are gain controls, a feature typical of pro amps. Gain controls are useful in a number of studio situations. For example, near-field reference monitors are often too loud for comfort when the gain on your mixer or preamp is set within its most effective operating range. Rather than turning down the mixer, which will affect the signal-to-noise ratio, you can reduce the volume with the gain controls on the amplifier.

All of the units in the test group have individual gain controls for each channel. The Carver's and the QSC's controls are mounted on the rear panel, and the Hafler's and BGW's are on the front. Rear-mounted controls make sense in touring and sound-reinforcement situations, where you don't want unauthorized personnel screwing things up. (The Carver's gain controls can be permanently bypassed by cutting a jumper on the main circuit board.) Front-mounted controls make more sense in the studio, where you want things handy. The Hafler comes with round plastic plugs that can be inserted over the gain controls to make them "tamperproof."

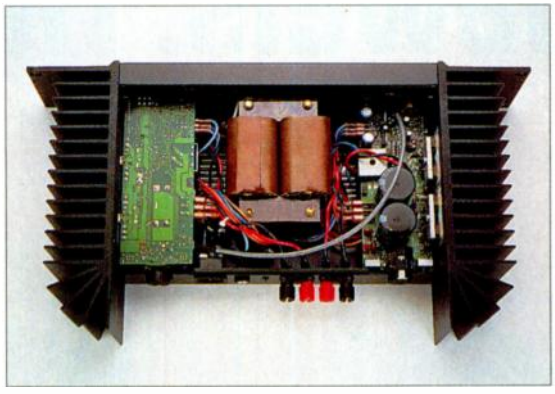


FIG. 2: The Hafler P3000 disperses heat via large, distinctive, cast-aluminum heat sinks, which include the sides of the unit, the rack ears, and several inches of the front panel.

The gain controls on the Hafler and QSC are continuously variable; those on the Carver and BGW are detented, moving in "clicks." However, the Carver's controls can be set between detents, which indicates the use of a continuous potentiometer (variable resistor) similar to those used in the Hafler and QSC with detents added. *Spinal Tap* fans should note the number of detents provided: this amp goes to 11!

The BGW's gain controls, on the other hand, are actually *stepped attenuators*. This means each detent (there

SPEC-ULATION

When shopping for a new car, I doubt you would base your decision solely on the manufacturers' printed specifications. Sure, it's good to know a vehicle's specs: the number of valves per cylinder, 0-to-60 time, top speed, braking distance, front leg room, and so forth; all are important criteria. But by themselves, these objective facts tell you little or nothing about whether you will enjoy driving a particular car for the next ten years or so.

The same thing holds true with power amplifiers, yet many people insist that amps with similar specs sound the same. Of course, if that is true, then any amp that measures *better* than another must necessarily sound better, as well. Although this would make shopping easy—the amp with the best specs wins—it doesn't work that way in the real world.

The specs in the table below were submitted by the manufacturers and are all based on the same measurement criteria. (Unless you know exactly how a given spec was measured, it is meaningless and should not be used for comparison purposes. Because each manufacturer involved in the test seemed to spec their amp differently, I sent each a form requesting information based on standardized criteria.)

If the amps were ranked strictly according to these specifications, the BGW would come out on top. It has the best measurements overall, with the lowest THD and intermodulation (IM) distortion, highest signal-to-noise ratio, excellent damping factor (an amp's ability to stop the speaker cone from moving, which affects low-end response), and slew rate (a measurement of how quickly an amp can transition from a positive-going wave to a negative-going wave, which affects its ability to reproduce transients).

The Hafler would come in second; its power bandwidth and slew rate are exceptional, and its damping factor equals that of the BGW. A detailed explanation of these specifications is beyond the scope of this article. To learn more, I highly recommend the excellent *Yamaha Sound Reinforcement Handbook* (available from Mix Bookshelf; tel. [800] 233-9604 or [510] 653-3307; fax [510] 653-5142).

The QSC would follow very closely behind the Hafler. Although many of its specs are excellent, particularly its 110 dB signal-to-noise ratio, it has a much lower slew rate and damping factor. The Carver would rank last, held back by its relatively low signal-to-noise ratio and slew rate.

Ironically, the panelists—who were unaware of the specs until after the test—ranked the amps in pretty much the opposite order! The conclusion is obvious: as stated earlier, specs tell you very little about a power amplifier's real-world performance.

If specs are unreliable as a performance indicator and you can't conduct a proper listening test at the store, on what should you base your purchasing decision? Buy the amp from a reputable dealer that sells several brands and allows trade-in privileges. Take the amp home and give it a thorough test drive in your studio. Live with it for a week or so and judge its performance with the most accurate and sensitive measuring devices known: your ears.

	BGW Model 200	Carver pm420	Hafler P3000 trans•nova	QSC Series One Model 1200
Continuous Avg. Output Pwr. (into 8Ω; 20 Hz-20 kHz ±1 dB)	100W/ch. @ 0.08% THD	135W/ch. @ 0.2% THD	150W/ch. @ 0.1% THD	100W/ch. @ 0.1% THD
Continuous Avg. Output Pwr. (into 4Ω; 20 Hz-20 kHz ±1 dB)	150W/ch. @ 0.08% THD	210W/ch. @ 0.2% THD	200W/ch. @ 0.2% THD	150W/ch. @ 0.1% THD
Freq. Response (+0 dB, -1 dB* @ 1W, 8Ω)	20 Hz-20 kHz	20 Hz-20 kHz	0.15 Hz-200 kHz	20 Hz-20 kHz
Power Bandwidth **	20 Hz-20 kHz @ 100W, 0.8% THD	10 Hz-60 kHz @ 135W, 1% THD	0.15 Hz-100 kHz @ 150W, 1% THD	5 Hz-60 Khz @ 100W, 0.1% THD
Dynamic Headroom (4Ω)	2.0 dB	1 dB	2.0 dB	3 dB
Dynamic Headroom (8Ω)	1.5 dB	.5 dB	1.6 dB	2 dB
S/N Ratio (20 Hz-20 kHz)**	>103 dB unweighted	>95 dB A-weighted	>100 dB A-weighted	>110 dB A-weighted
THD (20 Hz-20 kHz @ 8Ω)	<0.08%	<0.1%	<0.1%	<0.1%
THD (Typical @ 1 kHz)**	N/A	0.05%	0.01%	0.01%
IM Distortion (SMPTE-IMD)**	<0.02%	<0.1%	<0.1%	<0.025%
Damping Factor (@ 8Ω, 20 Hz-1 kHz)	>400	>200	>400	>200
Slew Rate	40 V/µS	10 V/µS	100 V/µS	12 V/µS
Power Consumption**	400W	540W	700W	528W

* BGW: -0.25 dB ** Referenced to rated power into 8Ω.

are 22) switches a precision resistor into the circuit; there are no "in between" positions. Many purists feel that stepped attenuators sound better than potentiometers. In normal studio applications, these differences are trivial. However, they do make a big difference in situations where the gain of multiple amps must be precisely matched: for example, when biamping or performing the listening test we'll be conducting later.

JACKS ON THE BOX

Another sure-fire way to tell a professional amplifier is by its input jacks. No self-respecting pro amp would be caught dead with unbalanced, "consumer" RCA inputs on its back panel. All amps in this group except the QSC have balanced tip-ring-sleeve (TRS), ¼-inch inputs that can also accept unbalanced connections.

The QSC does have balanced ¼-inch input jacks. However, instead of standard TRS wiring, they are configured as RTS (ring-tip-sleeve) jacks. In other words, their polarity is reversed with respect to the speaker output. Unless this is rectified, the QSC will be out of phase with just about every other amp on the market, including the other three in this test. Fortunately, there's an easy cure: you simply reverse the speaker outputs, connecting the red terminal to the speaker negative and the black terminal to the speaker positive.

Why did QSC do it? According to QSC Senior Scientist Paul Ierymenko, inverting the input signal with respect to the output prevents "high-frequency, radiated feedback that might occur if a user bundles the signal wires with the speaker wires when he or she is dressing the rack. It's a little more of a bulletproof situation; the feedback becomes negative, and you're less likely to have a system oscillate." Perhaps so. However, as Ierymenko pointed out, this technique dates from QSC's early days, before many of our current standards were established. In today's marketplace, where users expect the tip to be positive and the ring negative, going against the flow no longer makes sense. QSC seems to agree: Ierymenko told me the company has already gone "tip positive" on their new PowerLight series, and this practice will be carried out on all future models.

In addition to the ¼-inch jacks, the BGW and QSC provide separate XLR inputs. (The QSC's XLR jack is wired conventionally, with pin 2 hot.) The Hafler uses two-in-one connectors that accept either XLR or ¼-inch plugs. The QSC throws in a screw-terminal "barrier strip," which is unlikely to see much service in studio applications.

All of the amps except the BGW have identical heavy-duty, 5-way binding posts that are spaced for use with dual banana plugs. The BGW also has 5-way posts, but they are spaced in such a way as to preclude the use of dual bananas in normal stereo hookups. (They accept dual bananas only in the mono-bridged configuration, in which only the two red terminals are connected to the speaker.)

▼ **The power amplifier is one of the most important components in a studio.**

PROTECTION RACKET

All of the amps have various built-in amplifier/speaker protection circuits that activate in the event of catastrophe. All four are protected against overheating with thermistors that continuously monitor the temperature of the heat sinks on each channel. If the temperature exceeds a preset point, the channel is shut down until it cools, at which time it comes on again automatically.

All four are also protected from short circuits. The Carver and BGW both accomplish this with circuits that monitor the amount of current passing through the output transistors, shutting down the amp in the event of a sustained high current draw such as that caused by a short.

QSC takes a different approach, using a patented Output Averaging circuit that continuously monitors the load impedance. As the impedance drops below 2 ohms, the circuit progressively limits current to a level the output transistors can handle. Hafler also takes a different tack, using an output-short detector that compares the input and output voltages on each channel, shutting down the affected channel in the event of a short. Turning the amp off and on again resets the circuit. Of course, each manufacturer claims it's protection scheme is the most transparent.

One thing you never want to see coming out of an amp is direct current. Small DC voltages will offset the speaker cone from its usual center resting position. If the voltage is large, however, as can happen in the event of an output transistor failure, the speaker's voice coil will probably go up in smoke. All of the amps except the Hafler have some form of DC-offset protection to prevent such disasters.

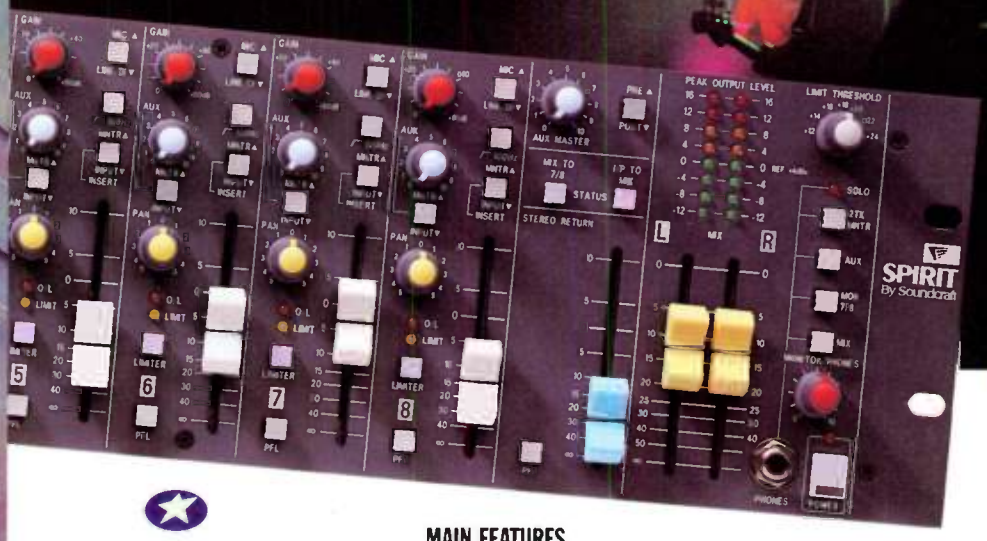
A few remaining features round out the amps' protection schemes. All four protect the speakers with a turn-on/off muting circuit. The Carver has a clipping eliminator that detects hard clipping and reduces the input signal to compensate; purists can disable it by cutting a jumper on the main circuit board. The QSC's inputs are protected against damage from extremely high input signals or RF interference, and the outputs are stable into reactive or mismatched loads. Finally, if everything else fails, all the amps but the Hafler include an AC circuit breaker; the Hafler has a line fuse, instead.

The amps communicate their operating status with a variety of LEDs. All four have indicators for power (the Hafler's is on its power switch) and clipping. On all but the Hafler, the clip light also illuminates to indicate that one of the amp's protection features has activated; the Hafler has separate clip, thermal, and short indicators. Last, but not least, all of the amps except the QSC also have signal-present lights.

SHUT UP AND LISTEN!

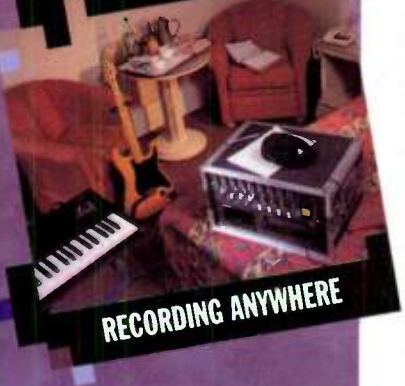
The four amps in our test group are obviously well equipped to handle the

MIX TO DIGITAL MULTITRACK ANYWHERE



It's so easy to take a digital multitrack recorder out on the road. But teaming it up with a compact, affordable mixer that's up to the job of making studio quality live digital recordings direct to tape is not so easy.

So Spirit have created ProTracker.



We've packed a high quality 8 channel in-line mixer into a 3U case. ProTracker has an ultra clean signal path, and everything you need to make professional digital quality recordings – yet ProTracker will also excel when you want to overdub; it'll even mix, your front-of-house sound while you're recording. ProTracker's key to digital quality recording lies in its unique integration of superior components into the straightest possible signal path between its input and its balanced tape output, effectively giving you a pure direct-to-tape mix. To keep a tight rein on levels going to tape – and to prevent accidental digital overload distortion when a band member decides to crank up the volume – there's a fast limiter with switchable threshold: you just set and forget it. Take a good look at the impressive feature list...

MAIN FEATURES

- In-line monitoring signal format – 2 discrete inputs per channel
- 8 channels with 60mm linear faders (>90dB cut off)
- Expansion sockets for daisy-chaining ProTrackers
- New high quality, high gain mic pre-amp (-129dBu EIN) with switchable 48V phantom power on every input
- Balanced Line inputs with built-in DI capability for instruments
- Switchable High Pass Filter on every channel
- Built-in Limiter (300µSec attack time/ 3 Sec release), selectable on every channel
- Insert and Aux switchable between channel and monitor paths
- Aux globally switchable pre/post fader
- PFL on each channel
- Overload and Limiter indicators on each channel
- Monitor fader and Pan control
- Balanced Tape Send/Return, switchable between +4dBu and -10dBV
- Separate pre-fade Insert and Return sockets, eliminating Y-cables
- Stereo Effects Return with fader and PFL
- Inputs switchable to Mix to allow simultaneous front-of-house mixing and recording
- Mix routable to Tape Sends 7/8 for simultaneous 2-track recording, without affecting multitrack feeds from channels 1 to 6
- Headphone monitoring of 2-track return, Aux, 7/8 or Mix
- Solo indicator
- Monitor Outputs follow headphone output
- Mix Output and 2-Track Return accept +4dBu XLRs or -10dBV RCA phonos
- 100 – 240 VAC operation with internal universal switching power supply
- Road quality construction

See us at AES Booth #707

Superb Live Recordings

ProTracker is the perfect live recording partner for affordable digital multitracks such as ADAT* and DA-88**; and, like digital multitracks, you can link several ProTrackers together for additional tracks. Each channel is equipped with our best-ever mic preamp and a balanced line input with direct instrument capability, so you don't have to use any noisy DI boxes. A High Pass Filter on each input cuts mic popping and stage rumble. With ProTracker, you can not only record to multitrack but you can also make and monitor a 2-track DAT master at the same time.

Overdub or Record Anytime, Anywhere

ProTracker is designed for the road. That's why we have incorporated a power supply that works all around the world, so you can make the best location recordings wherever you go. And with its compact rack design, ProTracker will fit into any outside broadcast or location recording system.

8760 South Sandy Parkway,
Sandy, Utah 84070.
Tel: 801-566 9135
Fax: 801-566 2086

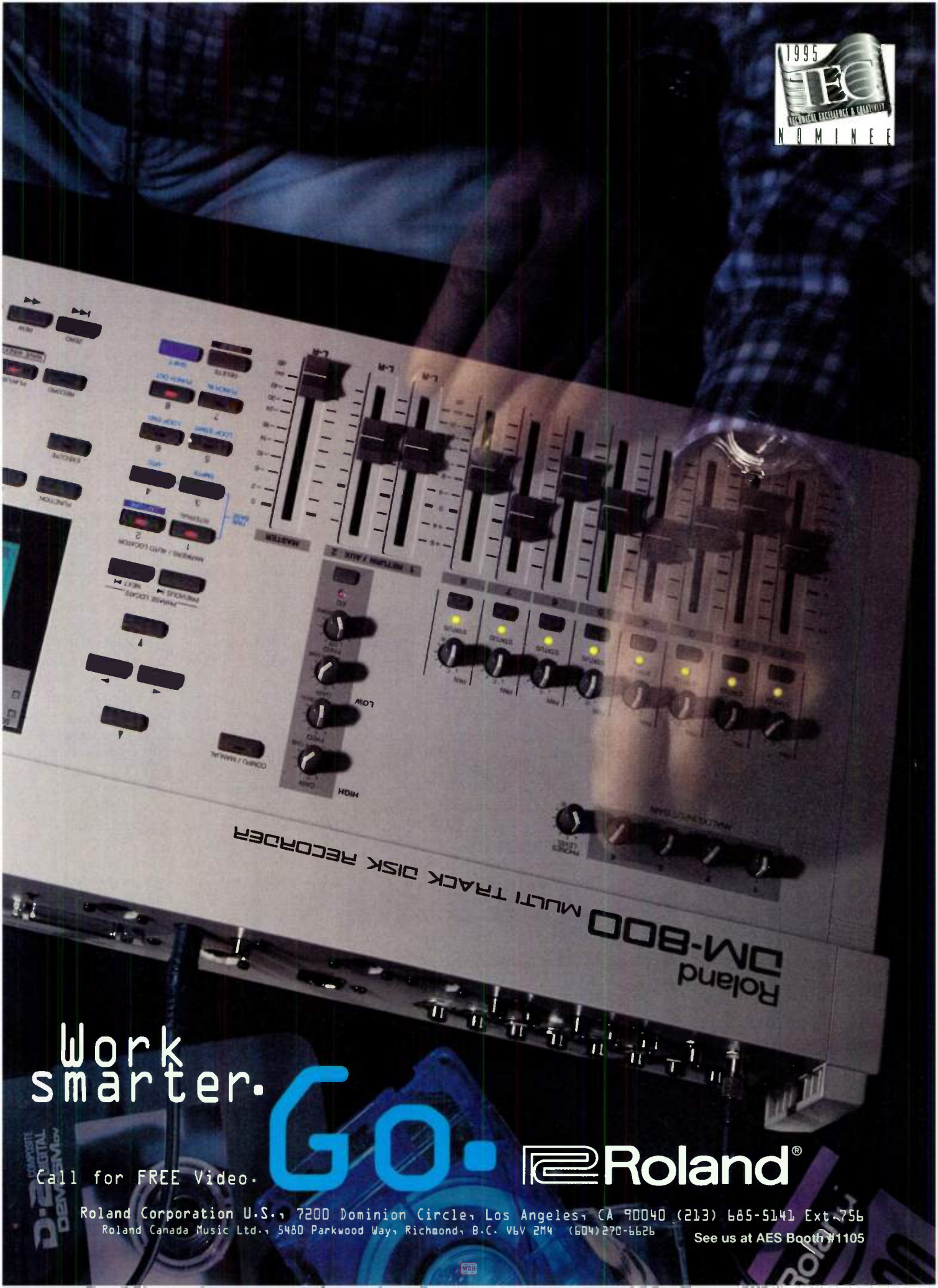
* ADAT is a registered trademark of Alesis Corporation
** DA-88 is a registered trademark of Tascam

SPIRIT
By Soundcraft

H A Harman International Company

pro TRACKER


$6295*
Powerful.
100 layers per track.
Bar/Beat Editing.
Time compression.
Pitch Correction.
Full Automation.
Interface for
DA-88/ADAT/RS-422.
Trigger Mode.
Portability.
No computer needed.
SMPTE/MMC/MTC/SPP Sync.
Video Output.
Scrub Preview.
Backup to SCSI/Audio DAT.
Non Destructive Editing.
Punch In/Out.
Reliability.
Portable/Stand Alone.
"No Crash" performance.
Fast.
Easy to learn.
Confidence.
DM-800
Digital Audio
Workstation.
*MSRP 4-1-95. Price subject to change.


1995
TEC
NOMINEE
Roland
DM-800 MULTI TRACK DISK RECORDER
Work smarter.
GO.
Roland®
Call for FREE Video.
Roland Corporation U.S., 7200 Dominion Circle, Los Angeles, CA 90040 (213) 685-5141 Ext. 756
Roland Canada Music Ltd., 5480 Parkwood Way, Richmond, B.C. V6V 2M4 (604)270-6626
See us at AES Booth #1105

rigors of studio life. However, all the fancy features in the world are worthless if the amp doesn't sound good. There's no getting around it: the only way to determine how an amplifier sounds is to listen to it. And here we enter into an area fraught with controversy.

The topic of amplifier sound quality has been hotly debated for years, with opinions falling into two diametrically opposed camps. Objectivists insist that any sonic differences detected between properly operated amps with equivalent specifications are either a figment of the listener's mind, a result of faulty test conditions, or both. Subjectivists are equally adamant that as long as you know what to listen for, each amp sounds quite different, no matter what the specs may say.

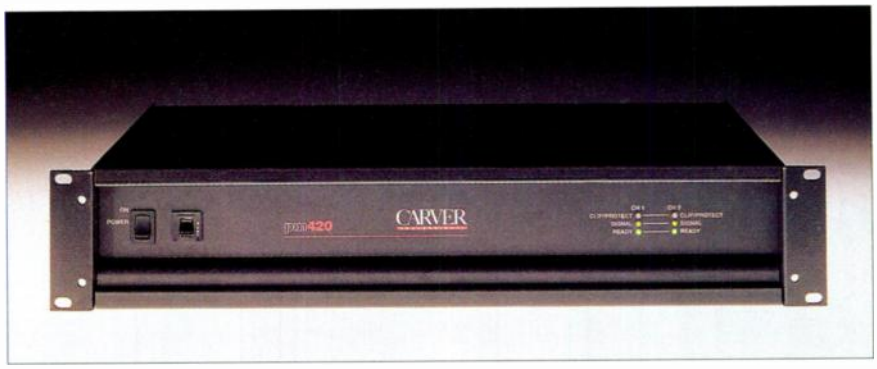
Experience conducting numerous comparative product tests of consumer-electronics components—I've compared groups of receivers, VCRs, surround processors, projection TVs, speakers, and many other devices—has proven to me that both sides speak some truth. Time after time, no two products—even those with closely matched specs—have sounded or looked exactly the same. Given the different design variables involved, how could it be otherwise? Of course, the perceived differences are often extremely subtle, and whether they make one product better or worse than another is entirely subjective.

On the other hand, unless a listening test is set up carefully, differences may be heard that don't really exist. In particular, the testing environment must form as level a playing field as possible. Only the device under evaluation must change; every other variable must stay the same.

In the case of power amplifiers, this means each amp must receive exactly the same input signal and feed exactly the same pair of speakers. Because people tend to perceive a louder device as sounding better, each amp's output level must be precisely matched within a fraction of a dB. And because human aural memory is short, you want to be able to switch between amps as rapidly as possible. Finally, to prevent any conscious or unconscious bias from skewing the results, it's a good idea if the persons doing the evaluation are "blind," meaning they do not know which amp is which during the test. (Obviously, because I had to organize and conduct the proceedings, I knew which amp was which. Technically, that makes this a single-blind test.)

Although I planned at first to feed the amps from a mixer, I quickly realized that this would make it extremely difficult to ensure each amp was receiving an identical input signal. Instead, I used an Acurus A10 line preamplifier, which is a no-frills, high-quality device with two pairs of unbalanced RCA line outputs. Each output was then split into two with a Y adapter. All output cables were identical 2-meter audio interconnects from Monster, each terminated on the amp side with a ¼-inch TS adapter.

To allow the four amps to be rapidly switched between a single pair of speakers, I used a ProCo RMS 2A switcher. Actually designed to direct the output of a single amp into several pairs of



The Carver pm420's recessed front panel and solid cabinet makes it well suited for life on the road.

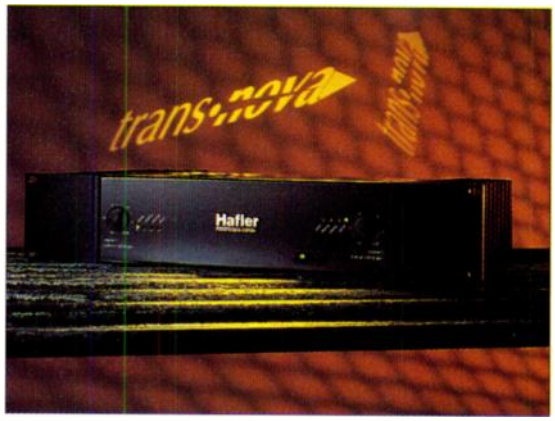
monitor speakers, the RMS 2A is a passive, 1U box with four toggle switches on the front and an impressive row of 5-way binding posts on the back. I simply used it in reverse, feeding the speaker output of each amp into one of four ALT positions: 1A, 1B, 2A, and 2B. The output of the box was connected to the monitor speakers.

All of the speaker connections were made with identical 8-foot lengths of high-quality, finely stranded, 14-gauge speaker wire. To ensure that each connection was solid and identical, I terminated each cable with ITT Pomona dual banana plugs. These also made it easy to verify correct polarity, not a trivial concern given the large number of connections involved.

LISTENING LABORATORY

After clearing my long-suffering and wonderfully understanding wife and daughter out of the house for the day, I rearranged the furniture and set up the equipment in our 24 × 13-foot living room/home theater. (My bedroom studio is much too small to comfortably accommodate four amplifiers, several pairs of speakers, and four people!)

The main speakers used for the evaluation were B&W Matrix 801 Series 2 Anniversary Editions. (The current Series 3 version sells for $5,500/pair.) Popular in major studios worldwide for over a decade, these high-resolution professional monitors are a 12-inch, 3-way design. Rated at 8 ohms nominal, with a moderate sensitivity of 87 dB at 1 meter for 2.83V, the big B&Ws present a fairly easy load to the amplifier; their power-handling range is 50 to 600 watts.



The 150W/channel Hafler P3000 produces power with MOSFET transistors and the company's trans•*nova* technology, rather than the usual Class AB construction and bipolar transistors.

The amps were arranged side-by-side on the floor, to one side of the room. The preamp, CD player, and switchbox were placed on a table in front of the listening position, which was located approximately nine feet back from the speakers. All input and speaker cables were carefully dressed to avoid interference, and everything was plugged into the same AC outlet. The amps were powered up and left on for several hours before being calibrated.

With so many amplifiers involved, I anticipated ground loops would be a problem. I was not disappointed: when first plugged in, all four amps hummed the same annoying 60 Hz tune. Fortunately, the BGW and Hafler both have ground-lift switches (a *much* appreciated

How in the world did people work before Pro Tools™?

[painfully]

Sure, you could record, edit, process, mix and master your next platinum record without 16 to 48 tracks of record and play. You may not need 64 channels of input and output. Or every sound-perfecting extra from our 100 Development Partners.

But the reason more top music professionals use a Pro Tools III system than all other digital audio workstations combined, is that it opens up more creative possibilities. Pro Tools III provides more options to manipulate vocals, drums and other instrumentation, in a random-access, non-destructive format. And, thanks to our TDM digital mixing and DSP Plug-In environment, you have complete control, all on one desktop, all in one high-quality digital format.

You can even burn a CD the moment you've completed your masterpiece.

> "Pro Tools is my Swiss army knife for recording."
>
> Jerry Harrison
>
> LIVE, Crash Test Dummies

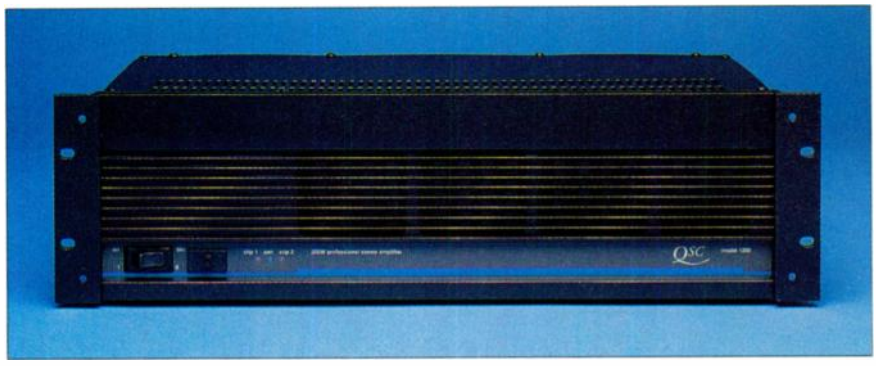
In short, Pro Tools III is all about making music better, not just faster. Which is exactly how you've wanted to work, all along.

To learn more, just call us at 1(800)333.2137, x173. We'll send you a free information kit.

digidesign

A · V · I · D

San Francisco • Paris • Seattle • Munich
New York • Chicago • Nashville • Los Angeles
London • Melbourne • Eindhoven • Milan



The QSC Series One Model 1200 has no external heat-sink fins; it relies on its thick, louvered front panel, instead.

feature) on their back panels; engaging these took care of those two amps. The QSC and Carver were silenced with ground-lift adapter plugs. As they say in the car commercials: *Don't try this at home!* Disconnecting, or "lifting," the ground pin on an amplifier or other device can have lethal results. If you can't use balanced connections, a "quasi-balanced" interconnect with the shield disconnected ("telescoped") at one end should solve grounding problems.

After checking and rechecking each connection, I used Delos' *Surround Spectacular* test CD to verify that each amp's stereo channels and polarity were correctly wired. (In a test situation such as this, no detail can ever be taken for granted.) With everything in position and correctly wired, I began the critical process of matching each amplifier's gain.

Setting the 315 Hz "System Setup and Balance Tone" on *The Sheffield/XLO Test and Burn-In CD* to infinite repeat, I used a digital voltmeter to measure the output voltage of each channel on each amp. With all gain controls set to max, the Hafler turned out to have the lowest output voltage, so I decided to adjust all the other amps to match it. Unfortunately, the detented gain controls on the Carver and BGW made this task easier said than done.

After an hour or so of frustrating knob fiddling, I got everything perfectly matched except for one small detail. With both channels set one detent below max, the BGW's left channel was perfectly matched to the other amps. Its right channel, however, was off by -0.10V. This is difference of 0.56 dB, which is theoretically inaudible but troublesome, nevertheless. A call to the company confirmed that the unit's stepped attenuators cannot be set between detents and that there is no trim pot inside the unit. I was also told that a 0.10 volt variance was within the tolerance of the precision resistors used.

After double-checking each amp's level one more time with the voltmeter, I used pink noise and a sound pressure-level (SPL) meter to check the actual sound pressure coming out the speakers. As I switched between the amps with the meter held steady at ear height, the needle never varied from a plenty loud but not excessive 85 dB SPL. Note that at this level, none of the amps' clip lights illuminated even on the loudest peaks. Satisfied they were now level-matched as closely as possible, the listening test was ready to begin.

TESTING, TESTING, 1-2-3

The listening panel arrived shortly after the setup was completed. It included Chris Meyer, Jeff Rona, and **EM**'s own Scott Wilkinson, a group with decades of serious listening experience. (See the "Lend Me Your Ears" sidebar for their biographies.) Although the panel members knew which brands and models were under test (the amps were not hidden from view), at no time did they know which amp they were actually

Worth its weight in gold–*and platinum.*

Tony Brown discovers the AT4050 studio microphone.

With 8 gold and 14 platinum records, 41 number-one singles, and 4 Grammy Award winning titles, record producer Tony Brown has a reputation for knowing excellence when he hears it. So it's no surprise that once Tony tried an AT4050 multi-pattern capacitor microphone, he's been using it ever since.

"I use it every session on something," says Tony. "There are several great mics that engineers always like to bring out–and this is one of them."

The Audio-Technica AT4050 delivers supremely transparent and accurate sound without sacrificing warmth and ambiance. The large-diaphragm design utilizes two capacitor elements to provide consistent, superior performance in three switchable polar patterns (cardioid, omnidirectional and figure-of-eight). Transformerless circuitry provides exceptional transient response and clean output even under extremely high SPL conditions.

And while it's certainly *worth* its weight in gold, the rich, versatile performance of the AT4050 is available for much less than $400 per ounce!

To discover studio microphone excellence for yourself, call, write or fax:

Audio-Technica U.S., Inc.
1221 Commerce Drive
Stow, Ohio 44224
(216) 686-2600
Fax: (216) 686-0719

U.K.: Audio-Technica Ltd.
Old Lane, Leeds LS11 8AG
0113 277 1441
Fax: 0113 270 4836

audio-technica®

NOW THERE'S A KEYBOARD THAT GIVES YOU PLAY-BY-PLAY COVERAGE.

Know the score with Wurlitzer's WX Digital Keyboards. They bring a whole new sense to music: you don't just hear the music, you see it.

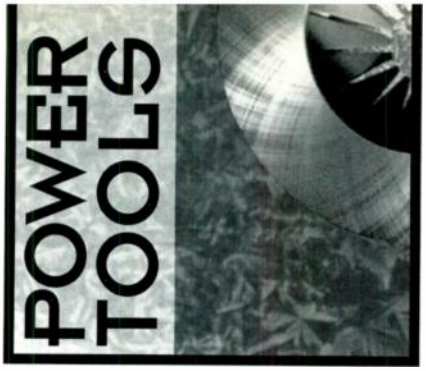
Multimedia capabilities make it possible to see the melody, chords and lyrics on a large, built-in screen, an external monitor or TV set. And the outstanding sound quality, exceptional sound selection, ease of operation and 16-track sequencer give you professional capabilities in your own home.

The WX2 packs all this into a portable keyboard. The WX400 offers the same advantages in a complete piano configuration, with 88 weighted keys. The WX Expander houses all these features in a desktop module that can be connected to any equipment with MIDI capabilities.

Call or write to hear more about the WX Series. And see what's playing now.

Wurlitzer DIGITAL

422 Wards Corner Rd. Loveland, Ohio 45140 • 1-800-876-2976



hearing. They also did not know any of the amps' specs, including power ratings and list prices, until after the test was complete.

During the test, the amps were identified only as 1A, 1B, 2A, or 2B, corresponding to the switchbox settings. The panelists did not know the actual assignments, of course, which were as follows: 1A was the Hafler P3000, 1B was the Carver pm420, 2A was the QSC Model 1200, and 2B was the BGW Model 200. These assignments were made at random when I set the amps

▼

Another sure-fire way to tell a professional amplifier is by its input jacks.

up, and they did *not* correspond to the amps' order on the floor.

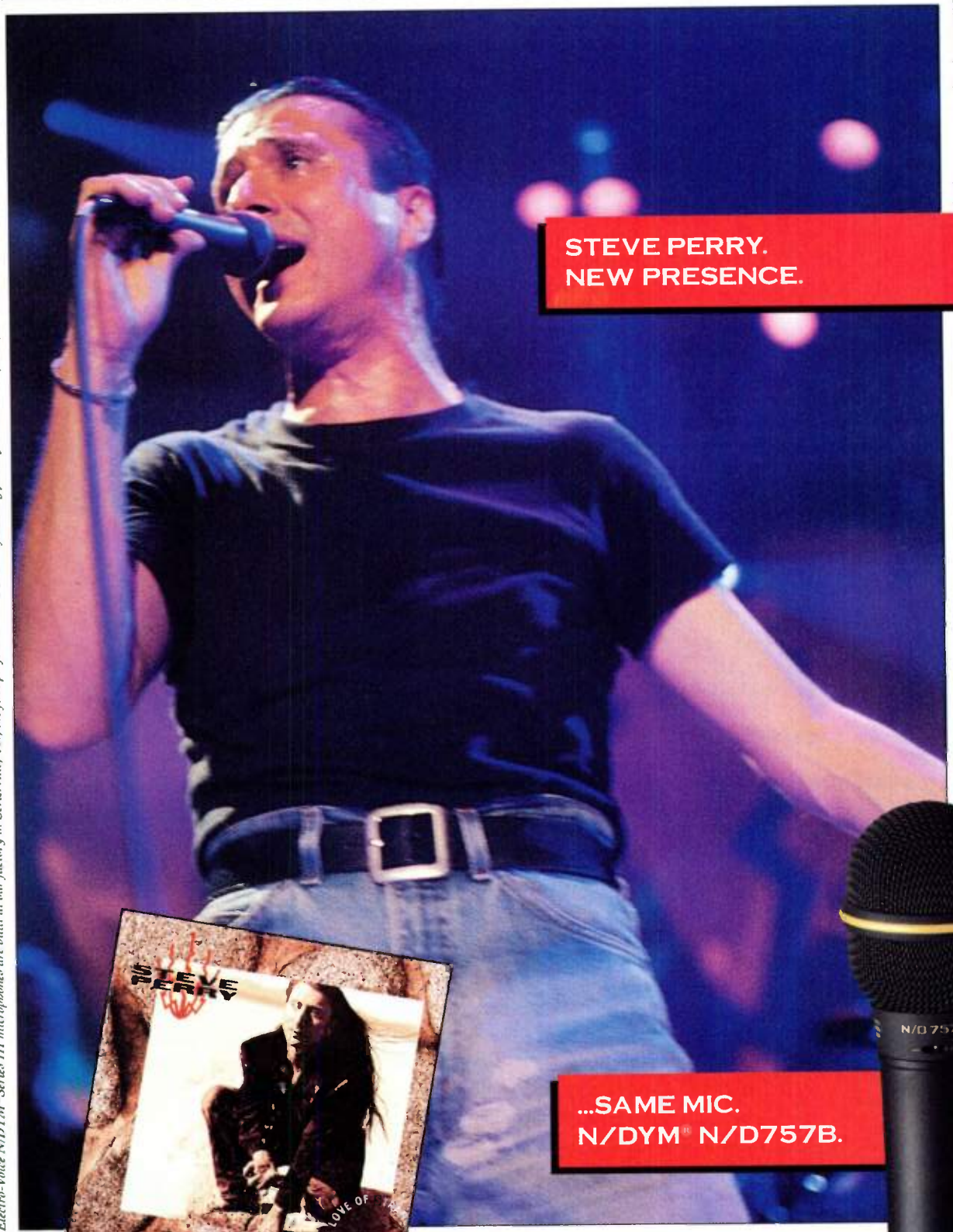
We began the test with "Red Dress" off Laurie Anderson's meticulously engineered *Strange Angels* CD. This track begins with a dry solo sax in a noisy club ambience, followed by Anderson's distinctive vocal sound. We also listened to several cuts from two CDs whose creation panelist Jeff Rona was personally involved in: Jon Hassell's highly complex *City: Works of Fiction* (on which Rona co-composed, mixed, edited, assisted in the mastering, and played keyboards and sampled percussion) and Hans Zimmer's soundtrack to the movie *Crimson Tide* (on which Rona contributed to the sound design and assisted in the mastering). After cleansing our palettes with broadband pink noise (useful for revealing frequency-response aberrations), we

assurance standard. No wonder N/DYM® Series III microphones offer the best sensitivity, highest output and the widest frequency response of any microphone in its class.

Electro-Voice, Inc, a MARK IV company

600 Cecil St. Buchanan, MI 49107 616/695-6831 800/234-6831 Canada: 613/382-2141

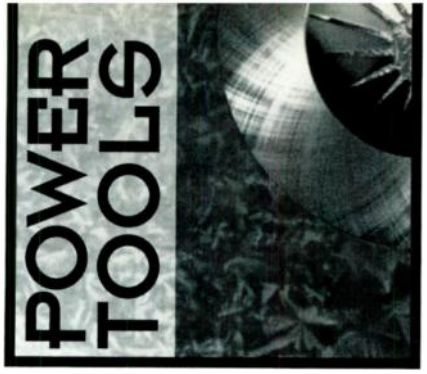
Electro-Voice N/DYM® Series III microphones are built at our factory in Sevierville, TN, the first professional audio manufacturing facility to receive the prestigious ISO 9002 certification — the world's foremost quality



"For the Love of Strange Medicine" — Columbia #CK44287

N/DYM® III — *Clarity and Power Defined.*

See us at AES Booth #812



rounded things out with two highly percussive cuts from Ryuichi Sakamoto's *Beauty* CD.

As we listened to each cut, I switched back and forth between pairs of amps (i.e., 1A/1B, 1A/2A, 2A/2B, etc.), not following any particular order, but making sure all were compared. Panelists were free to request any combination at any time but otherwise refrained from commenting about the amps until after the listening session.

FLYING BLIND

In a blind test such as this, industry reputation and conventional wisdom about which brand is "best" are removed as variables. Instead, the listeners are forced to base their opinions solely on what they hear, not on what they know about the amp from hearsay or personal experience. As you'll see, this can produce some surprises.

One thing should be made clear at the outset: these amps definitely do *not* "all sound the same." However, the differences are not glaring but are actually quite subtle. Asked to quantify the differences, Rona answered this way: "If I were to assign my favorite amp a ten on a scale from one to ten, my least favorite would be maybe a seven. They're all really close." Meyer agreed, noting that he would cluster his top three picks even more closely, as "the difference between a nine and a ten."

In fact, unless your monitor speakers are of sufficient quality, these subtle differences may be masked. In the days leading up to the test, I auditioned the amps using a variety of speakers, including Tannoy System 10 DMT IIs, M&K S80s, and Spirit Absolute 2s. None of these proved as revealing as the bigger, more expensive B&Ws.

Nevertheless, I wanted to give the panelists a chance to hear whether the amps responded differently to a smaller, more "real world" speakers. So after listening to the various CDs through the B&Ws, I placed a pair of Yamaha NS-10s on stands slightly more than one meter away from the listening position and repeated several of the cuts. The ubiquitous NS10s actually obscured the differences between amps and produced little useful commentary.

When all the panelists agreed they had heard enough, I turned off the amps and turned on my tape recorder. What follows is a summary of their comments. For the sake of clarity, I've substituted brand names in place of the numerical designations used by the panelists until after the test was over.

YOU SAY POTATO, I SAY...

In general, there was a remarkable degree of consensus about how each amp sounded. Rona pointed out that everyone seemed to have heard "75 to 80 percent the same thing. We just have different subjective judgments on what we find pleasing." Meyer concurred, noting that "when we talk about objective things like where we think the frequency bumps are, we all pretty much agree. But I think we have different subjective judgments on whether we *like* where those bumps are."

All three panelists remarked on the decided difference between the "Ones" (amps 1A and 1B; the Hafler and Carver, respectively) and the "Twos" (amps 2A and 2B; the QSC and BGW). This correlates perfectly with the amps' power ratings: both amps in the "Two" group are rated at 100 watts; those in the "One" group at 150 watts and 135 watts. However, none of the panelists ranked the amps strictly according to power rating, as you might expect.

There was general agreement that the "Twos" sounded very similar and lacked low-end punch compared to the "Ones." Rona expressed it this way: "The QSC and BGW tended to always sound pretty much the same to me; in general they had less low end than the Hafler or Carver."

Although each panelist found different attributes to like and dislike about each amp, the Carver and Hafler provoked by far the strongest responses. For example, all of the panelists commented on the Carver's solid low end. According to Rona, "The Carver was the punchiest. I heard a significant difference between it and the other amps in transient response on the bass end, especially on kick drums and the 'big boom' [at the beginning of a cut from *Crimson Tide*]." Wilkinson agreed, noting that he thought "the Carver was much crisper than the others in the low frequencies."

However, although Rona and Meyer agreed that the Carver's midrange response was a bit forward, or "pushed," compared to the other amplifiers, they disagreed on whether or not this was good. Rona considered it a plus, saying he felt it made the Carver seem "louder and more present than the others." Meyer, on the other hand, "marked the Carver down because of its fat midrange, but I listen to 2-way speakers all day long, which tend to be somewhat more recessed in the midrange than 3-way designs [such as the 801s] are."

Hafler amps are widely praised for their ability to present a well-defined

LEND ME YOUR EARS

The members of the listening panel all make their living creating and listening to music through speakers and power amplifiers.

Chris Meyer has been involved in the design of products for several musical-instrument manufacturers, including Roland Corporation, where he is currently Manager of Technical Research. He also composes and creates audio for desktop graphics with his own company, CyberMotion.

Jeff Rona is a composer and synthesist in Los Angeles. He created the scores for the award-winning TV series *Homicide: Life on the Street* and *Chicago Hope*. He has done musical sound design for composers such as Hans Zimmer, Mark Isham, Basil Poledouris, and many others.

EM's Technical Editor Scott Wilkinson has been working in the electronic-music industry for almost ten years, first as a product specialist at Roland and then as editor of *Music Technology* and *Home & Studio Recording*. He has been a professional musician for over twenty years, playing such diverse wind instruments as euphonium, sackbutt, digeridoo, krumhorn, and conch-shell trumpet.

"Version 6 is a killer" EM Jan.95

ELECTRONIC MUSICIAN 1995 EDITORS' CHOICE

NEW! **UPGRADE TODAY!!!** WINDOWS 95 FRIENDLY

BAND-IN-A-BOX™

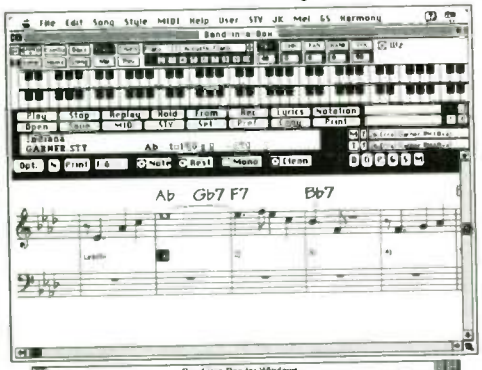
PROFESSIONAL • VERSION 6.0

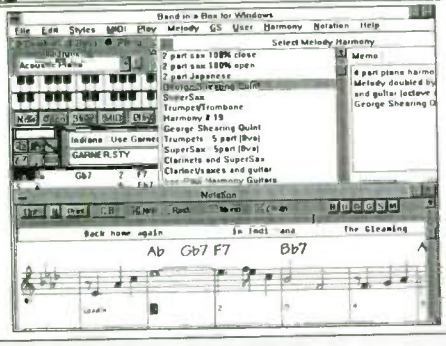
INTELLIGENT SOFTWARE FOR IBM (DOS/WINDOWS), MAC & ATARI*

(NOTE: ATARI Band-in-a-Box available only in Version 5)*

Version 6 for Windows and Macintosh is here. Automatic Accompaniment has arrived!

Type in the chords to any song, using standard chord symbols like C or Fm7b5, choose the style you'd like and Band-in-a-Box does the rest... Automatically generating professional quality five instrument accompaniment of bass, drums, piano, guitar & strings in a wide variety of styles.





100 STYLES INCLUDED WITH PRO VERSION. *Jazz Swing • Bossa • Country • Ethnic • Blues Shuffle Blues Straight • Waltz • Pop Ballad • Reggae • Shuffle Rock • Light Rock • Medium Rock Heavy Rock • Miami Sound • Milly Pop • Funk • Jazz Waltz • Rhumba • Cha Cha • Bouncy 12/8 Irish • Pop Ballad 12/8 • Country (triplet) • and 75 more!*

BUILT-IN SEQUENCER ALLOWS YOU TO RECORD OR EDIT MELODIES.

BUILT-IN STYLEMAKER™. *You can create your own 5 instrument styles using the* ***StyleMaker*** *section of the program.*

SUPPORT FOR OVER 70 SYNTHS BUILT-IN. *Drum & patch maps included for over 70 popular synths. General MIDI, Roland GS & SoundBlaster soundcard support included.*

NEW! Additional features in Windows/Mac Ver. 6

Band-in-a-Box 6.0 for Windows & Macintosh breaks new ground with over 50 new features including...

STANDARD MUSIC NOTATION *and leadsheet printout of chords, melody and lyrics. Enter your songs in standard notation & print out a standard lead sheet of chords, melody and lyrics.*

AUTOMATIC HARMONIZATION. *You can select from over 100 harmonies to harmonize the melody track, or harmonize what you play along in real time. Play along in "SuperSax" harmony, or harmonize the melody with "Shearing Quintet". Create your own harmonies or edit our harmonies.*

PLUS 50 MORE NEW FEATURES

OUR CUSTOMERS LOVE THE VERSION 6 FOR WINDOWS/MAC UPGRADE

"Wow !!... Version 6.0 is marvelous... I love the notation and harmonies... This is so much fun... You've added everything I wanted... The lead sheets look great... Bravo!... Congratulations"

"Band-in-a-Box is an amazing program"
Keyboard Magazine

"I am in awe. I didn't think that such an incredible program could even exist. This software is a dream come true."
PC Journal

– FINALIST –
PC Magazine Award
for Technical Excellence

After Hours Entertainment
Band-in-a-Box - PG Music

DownBeat – the #1 Jazz Magazine says...
"Band-in-a-Box is the most significant contribution to Jazz Education since Jamey Abersold Records."

BAND-IN-A-BOX PRICES

NEW CUSTOMERS (IBM/Mac/Atari*)

Band-in-a-Box PRO $88
(BONUS! IBM version now includes both Windows & DOS version for the same price!)
(NOTE: ATARI Band-in-a-Box available only in Version 5)*

ADD-ONS

Styles Disk #4 ..... $29
Styles Disk #5 *(Note: included with Version 6 upgrade)* $29
MIDI-FakeBook *(100 songs on disk)* $29
SUPER PAK *(Pro version + all 3 add-ons)* $147

UPGRADES

"Regular" Upgrade PAK to version 6.0 for Windows or Macintosh ... $49
includes version 6.0 upgrade + new Styles Disk #5, and Harmonies Disk #1. Order this if you already have the IBM or Mac PRO version 5.

"Complete" Upgrade PAK to version 6.0 for Windows or Macintosh .. $69
In addition to the regular upgrade PAK, this includes the 100 styles in the PRO version, and Styles Disk #4. Order this if you have an older version of Band-in-a-Box or a "bundled version", or are crossgrading (i.e. switching computer platforms).

MEMORY REQUIREMENTS: DOS (640K), Windows (3mb), Macintosh (4mb), Atari (1040)

HELP! *I Forgot to send in the Registration Card, but I want to upgrade now!!*
No problem. Since the upgrade checks for any previous version of Band-in-a-Box, you can order the upgrade even if you forgot to register!

Open LateNite
9 am - midnite
EST
7 Days

PG MUSIC INC. *Maker of PowerTracks and The Pianist & Guitarist series*
266 Elmwood Avenue Suite 111 Buffalo NY 14222
Phone Orders 1-800-268-6272 *or* **604-475-2874**
e-mail orders: Internet:75300.2750@compuserve.com
VISA/MC/AMEX/cheque/mo/po# **Fax 604-658-8444**

Hot new software programs created by PG Music!

Windows 95 Friendly! | Band-in-a-Box Pro Version 6 | PowerTracks Pro Version 3.0 | The Pianist Volume 2 | Jazz Pianist Volume 2 | New Orleans Pianist | Ragtime Pianist

"Version 6 is a killer" EM Jan.95

ELECTRONIC MUSICIAN 1995 EDITORS CHOICE

NEW! ONLY $88 (upgrade $49)

BAND-IN-A-BOX

PROFESSIONAL • VERSION 6.0

INTELLIGENT SOFTWARE FOR IBM (DOS/WINDOWS), MAC & ATARI

Version 6 for Windows and Macintosh is here. Automatic Accompaniment has arrived!

Type in the chords to any song, using standard chord symbols like C or Fm7b5, choose the style you'd like and Band-in-a-Box does the rest... Automatically generating professional quality five instrument accompaniment of bass, drums, piano, guitar & strings in a wide variety of styles.

- *Built-in Sequencer allows you to record or edit melodies*
- *100 Styles included with Pro Version*
- *Built-in Stylemaker™ lets you make or edit styles*
- *Support for 70 Synths built-in*
- *Standard Music Notation & Leadsheet printout*
- *Auto Harmonization – create or edit harmonies*

PLUS... 50 MORE NEW FEATURES IN VERSION 6!!!

OUR CUSTOMERS LOVE VERSION 6.0

Wow!... I love the notation & harmonies. You've added everything I wanted... Bravo!"

The Jazz Guitarist™ (Windows, Mac, Atari) $49

A music program containing a huge collection of over 60 jazz standards, played on MIDI guitar by top jazz/studio guitarist Oliver Gannon

RECORDED IN REAL-TIME ON A MIDI GUITAR!
Hear the music with CD-quality through your sound card or MIDI system. Most pieces have bass/drums as well as guitar so you get a full sounding jazz trio for the tunes!

the jazz guitarist from PG Music Inc.

LEARN TO BE A GREAT JAZZ GUITAR PLAYER!
On-screen fretboard shows you exactly what notes & chords are being played on the guitar. Slow down the performance or, better still, step through the music chord by chord, so you can learn every note as it's played!

PLUS MANY MORE FEATURES...
Jazz Trivia Game & Guess That Song Game, Program Notes, Biographies (all on disk) • Over 60 Top Jazz Standards with Complete Guitar Arrangements • Listen to the music while you work in other programs • Special support for Roland GS or General MIDI Modules • Standard MIDI files can be copied & used in other programs or presentations • Use your existing sound card or MIDI synthesizer

NEW! Music Printout! ONLY $29 (upgrade $15)

Version 3.0

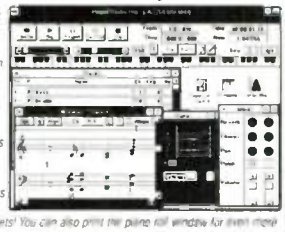
PowerTracks Pro™

SEQUENCER/NOTATION/PRINTING FOR WINDOWS (IBM)

"Solid sequencing at an unbelievable price" Electronic Musician Sept. 93

PowerTracks Pro 3.0 is a professional full featured MIDI sequencing, notation and printing program and is so easy to use! And we include versions for Windows 3.1 or Windows 95 AND DOS, so you'll be able to use PowerTracks PRO on all of your machines!

PRO RECORDING, PLAYBACK, SYNCH, EDIT & SYS-EX OPTIONS
48 tracks, real-step/punch record, sound on sound, MIDI File support, sync (SMPTE, Midi Time Code, MIDI), edit (quantize, cut/copy/paste, undo, data filters, transpose), multi-port support, 4Mb app limebase, sys-ex editor/librarian, patch names, banks & much more.



MUSIC NOTATION
Enter, edit, display music in standard Music notation. Intelligent automatic features such as: Correct beaming/tying of notes, reverse rests option, "Jazz Eighth notes" option (this automatically allows jazz swing eighth notes & triplets to be notated properly!). Reads in any MIDI File & displays it as notation!!

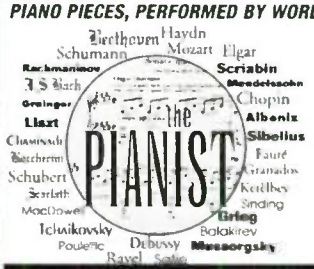
MUSIC PRINTOUT (ON ANY PRINTER!!)
Print any track in standard music notation. Selectable staves per page, and bars per line. Selectable margins and printer font. Portrait or Landscape (sideways) printing. Titles, composer, style, copyright information. Make your own lead sheets! You can also print the piano roll window for even more detailed analysis of a track.

NEWEST FEATURES...
✔Mixer for record & playback, lyrics, drum pattern editor, Piano, Auto Hand Splitting, ✔Programmable using DLL, ✔Patch/Bank names, ✔Non-GM Drum mapping, ✔Windows 95 friendly, ✔Over 30 new features in all (existing customers may upgrade for $15)

BUT POWERTRACKS GOES MUCH FURTHER... WITH EXCITING EXCLUSIVE FEATURES!
✔Enter, print out chord symbols in notation, ✔Automatic drum tracks (100 drum styles included), ✔Reads in chord symbols from Band-in-a-Box 6.0 MIDI files, ✔Patch caching for Gravis Ultrasound, ✔Comprehensive support for guitar on-screen guide, tab printout, ✔Built-in Roland SoundCanvas editor, ✔On-screen piano & guitar show notes as they're played, ✔Pro MIDI files included

POWERTRACKS FOR DOS VERSION INCLUDED FREE. *Yes! We include the DOS version for FREE in the same package. NOTE: The DOS version doesn't support music notation, or other graphical features.*

THE PIANIST... OVER 200 OF THE WORLD'S MOST POPULAR CLASSICAL PIANO PIECES, PERFORMED BY WORLD CLASS CONCERT PIANISTS! **$49**



Moonlight Sonata, Sonata Pathétique, Minute Waltz, Claire de lune, Mephisto Waltz, Hungarian Rhapsody, Fantasie Impromptu, Military Polonaise... & over 200 more!!!

PLUS... *Music Trivia Game, "Guess the Song", Program Notes, Biographies, Music Dictionary (all on disk) & much more.*

OUR CUSTOMERS LOVE THE PIANIST... *"Incredible... amazing... terrific... masterful... fabulous... love it... my favorite program!"*

NEW! The Pianist Volume 2

Volume II upgrade - $49 (requires The Pianist) • first time purchase: Volume 1 & 2 - $98

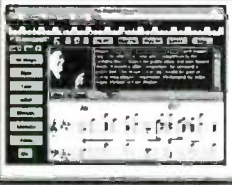
Turn your Pianist into a "SUPER PIANIST" with Volume 2!! We've recorded 200 more fabulous pieces for Volume 2. There are new program notes, new biographies and improved listings. If you thought the quality of the performances was great in the original Pianist program, just wait until you hear these new world-class, live piano recordings – you'll have endless hours of listening pleasure!

The New Orleans Pianist™ $49

Over 65 "New Orleans Style" piano music standards, played on MIDI keyboard by top New Orleans pianists Henry Butler, Jon Cleary, Tom McDermott, Joel Simpson & David Torkanowsky playing a wide variety of New Orleans, R & B, Blues & Ragtime piano music.

St. James Infirmary, When the Saints Go Marching In, Down by the Riverside, Burnt Mouth Boogie, Creole Lament, King Porter Stomp, The Pearls, Bogalusa Strut, My Bucket's Got a Hole In It, John Brown's Body, Margie, Charleston Rag, Maple Leaf Rag, The Entertainer, Raise the Rafters, Dirge for James Black and many more...

The Ragtime Pianist™ $49



Over 90 ragtime & early jazz piano standards, played on MIDI keyboard by top Ragtime Pianists... and featuring world-renowned Ragtime performer JOHN ARPIN!

Hear virtuoso performances of every Joplin rag in this program, as well as many other rags. CakeWalks, waltzes & other Ragtime Era tunes by Eubie Blake, Joseph Lamb, Daniels and, of course, Scott Joplin.
HEAR... *The Entertainer, Maple Leaf Rag, Chevy Chase, Easy Winners, Elite Syncopations, Fig Leaf Rag, Pineapple Rag, etc. (total of 34 Scott Joplin Rags), Indiana, Meet Me In St. Louis, St. Louis Blues, Bill Bailey, For Me and My Gal, & more...*

the JAZZ PIANIST from PG Music Inc. $49

THIS PROGRAM MAKES IT "TOO EASY" TO LEARN TO BE A GREAT JAZZ PIANO PLAYER!

Top jazz/studio pianists play 60 jazz standards in a wide variety of styles.
Hear the music with CD-quality through your sound card or MIDI system. Most pieces have bass/drums as well as piano so you get a full sounding jazz trio for the tunes! Jazz Trivia Game & Guess That Song Game, Program Notes, Biographies & Music Dictionary (all on disk).

NEW! The Jazz Pianist Volume 2

Volume II upgrade - $49 (requires The Jazz Pianist) • first time purchase: Volume 1 & 2 - $98

60 more fabulous jazz standards for Volume 2, complete with new program notes and biographies!

Sound Canvas Pro Editor/Librarian for Windows $29

Mixing/editing/storage of EVERY feature on Sound Canvas & other Roland GS & GM modules

You'll never need to touch your Sound Canvas front panel or open the manual again! Edit/design new sounds, new scales, tunings... YOU CAN EDIT EVERYTHING!!! Also plays MIDI files (hear changes as you make them).

Multi-MPU401 Driver for Windows $19

- *allows 8 music programs to use the same MPU401 at the same time*
- *a "must have" drive for all MPU401 Windows users (Roland, Music Quest, CMS, etc.)*
- *easy to use, installs as a driver in Windows to replace your current MPU401 driver*
- ***NEW!*** *Allows inter-program MIDI communication!*

REQUIREMENTS FOR ANY OF THE PROGRAMS:

IBM (DOS versions) *require 640K, MPU401/ MIDIATOR/SoundBlaster/SC-7, TG100*

WINDOWS (incl. Windows95) *versions require 3mb RAM+ any soundcard or MIDI interface*

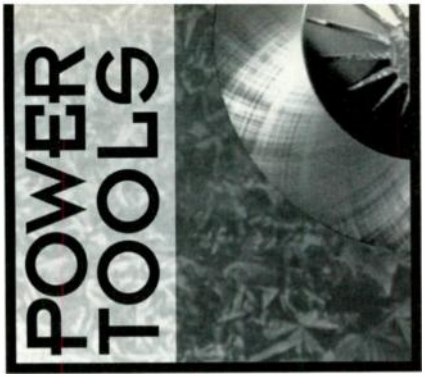
MACINTOSH *versions require 4mb RAM, system 6 or 7*

ATARI *versions require 1040ST or better*

Open LateNite 9 am - midnite EST 7 Days

PG Music Inc. *Maker of PowerTracks and The Pianist & Guitarist series*
266 Elmwood Avenue Suite 111 Buffalo NY 14222
Phone Orders 1-800-268-6272 or 604-475-2874
e-mail orders: Internet:75300.2750@compuserve.com
VISA/MC/AMEX/cheque/mo/po# **Fax 604-658-8444**
— Recorded Demo 604-475-2954 —

30 DAY UNCONDITIONAL MONEY BACK GUARANTEE ON ALL PRODUCTS



stereo image, and our blind test proved this to be well founded. All of the panelists agreed that the Hafler generated the best stereo soundstage. Rona noted that the Hafler "had a little bit better stereo image to me. It had a little more width than the other amps." A choral cut on the *Crimson Tide* CD provided an excellent test of stereo imaging, and here again he felt the Hafler "spread out" compared to the other amps. "Not only did I perceive more stereo image width," he said, "but I actually started to perceive the most amount of front-to-back image *depth*. When we switched to the QSC and BGW, however, the depth started to squeeze down a bit."

On the other hand, two of the three panelists had reservations about the Hafler's bass. Meyer called it "overemphasized," pointing out that, although "all of the amps seemed to have a bump somewhere in their frequency response, the Hafler's seemed the most radical, and it lies the lowest. This gives it a fat bass, but not in a good way. Wilkinson simply felt that the Hafler "wasn't as tight in the bass as the Carver or QSC."

The BGW provoked mixed reactions. Meyer liked this amp more than the other panelists, calling it "the flattest of the four. If anything, it had a slight recessed shelf on the high end. I did think it sounded a bit thin on some cuts, however. In my notes, I wrote 'nice transients, though not as tight as the QSC.'" Rona wasn't as impressed, characterizing the BGW's bass as "kind of wimpy and pinched." On the Sakamoto piece, which is a complex mix with plenty of percussion, he commented that it "sounded like the drums were having a hard time punching through." Wilkinson also found the BGW not to his taste. "I wrote 'muffled and muddy' in my notes. I just felt it was not as clean and open and present a sound as the other three."

Meyer and Wilkinson both found a lot to like about the QSC. According to Meyer, "The QSC was quite flat, with the least amount of 'flattery' [emphasis of one frequency range over another]. It also had the cleanest high-end transients, as well as a little bit better sense of 'air,' which for me means very low-level, high-end detail, such as room ambience. I thought it was tight in the bass, with sharp transients. Of course, my idea of a tight bass might be someone else's recessed or wimpy bass; it's subjective." Wilkinson—who owns a Model 1200—noted that, on the Laurie Anderson cut, the QSC had a "cleaner, tighter sound than the Hafler in the bass but seemed more closed in."

THE BOTTOM LINE

As noted earlier, blind tests can produce some surprises, often producing results that run contrary to prevailing popular opinion. Frankly, going into this test, I would have placed my bet on the Hafler. The P3000 is the most powerful amp in the group, and it costs more than the Carver or QSC. It also has great specs (see "Spec-ulation" sidebar). In addition, Hafler amps have received glowing reviews in other publications, including **EM**'s sister magazine *Mix*. I own a Hafler; I normally drive my 801's with a 250-watt/channel Model 9500 trans•*nova*. However, when asked to rank the amps in terms of overall sound quality, Rona and Wilkinson placed the amp they knew only as "1A" (the Hafler P3000) second; Meyer placed it last.

Instead, it was the Carver pm420 that emerged victorious, with Rona and Wilkinson placing it first, and Meyer putting it at third (after the QSC and BGW). This despite the Carver's lower list price, slightly lower power rating, and unremarkable specs. (Wilkinson tied the QSC and the Hafler for second place and ranked the BGW last; Rona placed the QSC third, followed by the BGW.) I am convinced that the results of this test would have been very different if the panelists had known the identity of the amps in advance.

Obviously, the panelists' opinions are entirely subjective and are based on listening to just two sets of speakers. Ultimately, it is extremely difficult to predict how a given amp will react to a given load. Some speakers, such as the B&W 801s used in our test, work well with a wide variety of amps; others do not. With different speakers, these four amps may well sound very different. Even when knowledgeable listeners agree on objective differences between amps, they will often disagree about whether these differences are good or bad. You must make the final judgment using your speakers.

The bottom line is that any of the four amps profiled here would do yeoman duty as a studio reference tool. All are robustly constructed, fully featured, and clean sounding. At 100 to 150 watts per channel, these particular models are best partnered with small, near-field reference monitors. If you have larger speakers or tend to monitor very loud and want greater headroom, you should consider moving up to a model with at least 200 to 250 watts per channel. But whatever you do, leave the consumer-grade amps to the people who will be buying your CDs. Your studio and your music deserve more respect than that.

Thanks to projects such as this, Associate Editor **Lawrence E. Ullman** *is spending the summer in front of his computer, deepening his already impressive studio tan.*

POWER AMP MANUFACTURERS

BGW Systems, Inc.
13130 Yukon Ave.
Hawthorne, CA 90250
tel. (800) 468-AMPS
fax (310) 676-6713

Carver Professional
PO Box 1237
Lynnwood, WA 98046
tel. (206) 775-1202
fax (206) 778-9453

Hafler Professional
546 South Rockford Dr.
Tempe, AZ 85281
tel. (800) 366-1619
fax (602) 894-1528

QSC Audio
1675 MacArthur Blvd.
Costa Mesa, CA 92626
tel. (714) 754-6175
fax (714) 754-6174

Korg changes the rules:

Classic synth sounds, physical models and total control can all live in one keyboard.

It's new. It's very new. The Korg Prophecy is here. With a system of synthesis based on our custom DSP and sound modeling technologies, Prophecy delivers a brain-jolting range of sounds – from classic analog synths to advanced physical models of acoustic instruments that capture every last nuance. It also gives you complete control with a pitch wheel, mod wheel and dynamic ribbon controller. So go break a few rules. Try out a Prophecy at your authorized Korg dealer.

Korg Prophecy

128 programs, plus a card slot that lets you easily save, load and access new sounds.

Pitch and mod wheels, ribbon controller and arpeggiator for complete expressive power.

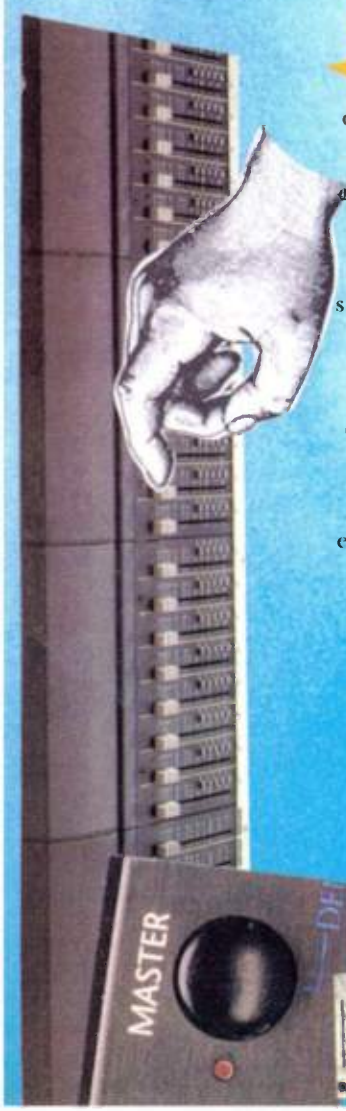
2 Resonant Multimode Filters for highest expressivity and real-time control.

7 dynamic effects, including distortion, wah, chorus, flanger, echo and more.

KORG® Prophecy
SOLO SYNTHESIZER

By Michael Molenda

Lip Service

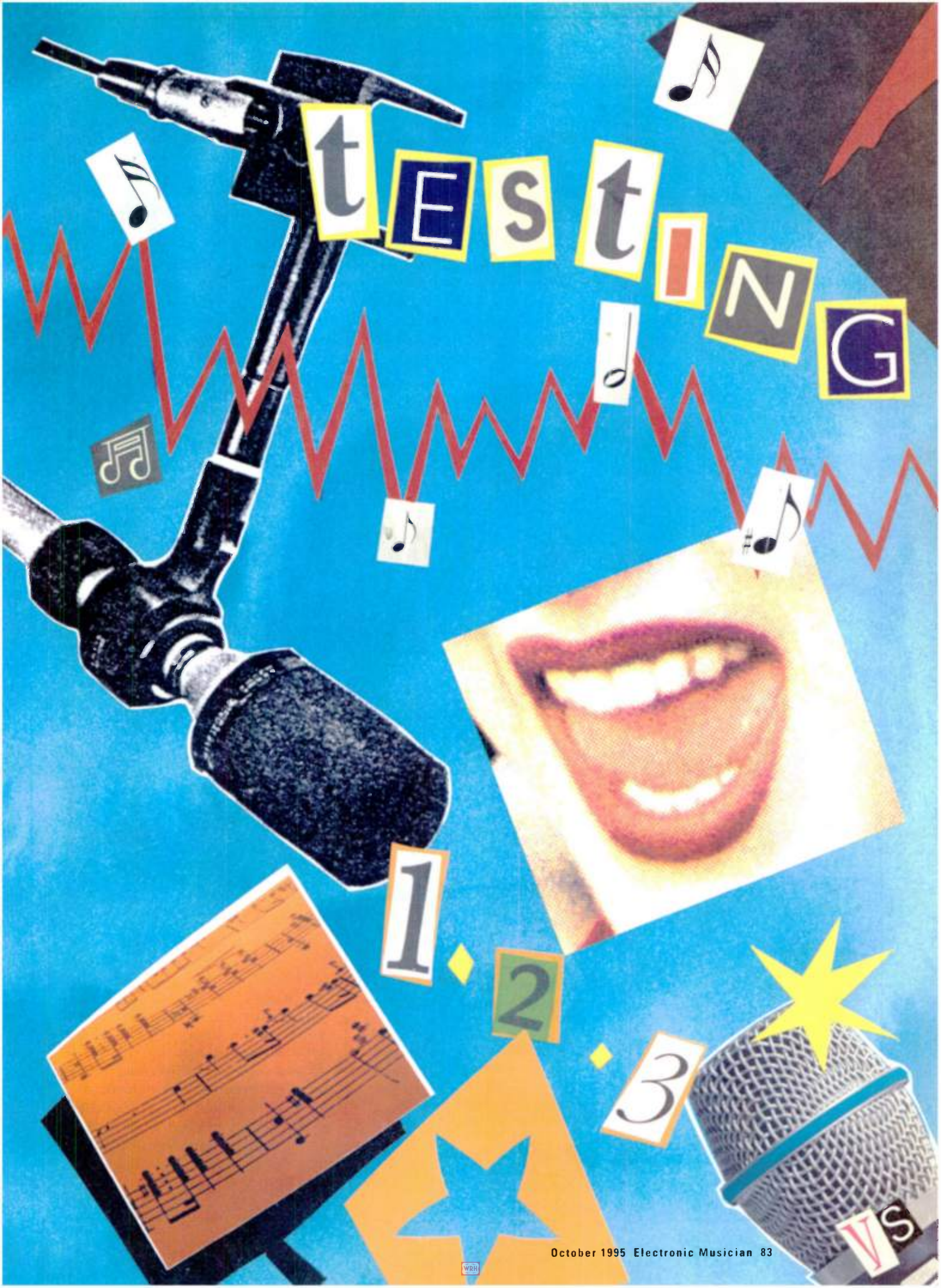


An expressive, achingly passionate vocal performance bursts from deep within a singer's heart and soul. If you're sitting behind a recording console when one of these little miracles occur, it's a truly wondrous experience—especially if the red "Record" light is on and you've captured the moment on tape. Unfortunately, an inspired voice can be a rare and fragile thing. It doesn't take much to kill a mood, throwing a singer into a place where it's difficult to deliver a competent performance, much less an awe-inspiring one.

The raw-nerved emotionalism of the typical vocalist can be a double-edged sword to the engineer/producer. When the sun and moon and stars are in perfect alignment, your track can be energized by a transcendent performance. However, a claustrophobic studio environment, a bad headphone mix, or a sonically unfriendly microphone can mess with a singer's concentration. And under the recording studio microscope, even minor annoyances can become huge problems; big enough, in fact, to sink an entire vocal session.

Illustrations by Valerie Spain

Tracking vocals is a delicate balancing act between emotion and technology.


testing
1.2.3
VS

LiP Service

So, if someone tells you that recording vocals is simply a matter of finding a microphone that enhances a singer's tone, you can be sure that person has never faced the sublime terror of a studio vocal session. Obviously, picking the right microphone is essential, but the best mic in the galaxy can't wring excellence from a demoralized, distracted, or downright angry singer. The real trick to recording beautiful (or ferocious) vocals is forging a harmonious balance between guts and intellect. In an ideal scenario, technology should never impede the artist's creative flow. Vocalists should be able to enter the studio and effortlessly bear their souls onto tape. And before you say, "Yeah, and I can pour gold coins from a bottle of Evian," there *are* ways to accomplish this feat.

First, the old Boy Scout motto "Be prepared" is of paramount importance. You need to have everything ready to go when the singer steps into the studio, and anticipate his or her needs throughout the session. Don't consider such acquiescence as pandering to a musician's "diva" complex; it's a blatant survival tactic for the recordist. Remember, a comfortable singer can really deliver the goods, while an uptight singer can make you wish you never sucked in breath when the obstetrician paddled your little fanny.

Second, your recording chops should be so well honed that the performer remains blissfully unaware of the technical requirements of the studio. The vocalist should not be forced to sing to the point of fatigue while you audition microphones, EQ tweaks, and compression settings. It's a shame how many engineers blow out their singers with incessant knob twiddling.

But now let's get down to specifics. I've broken up the vocal-recording process into separate sections, each loaded with helpful hints for maximizing the audio and emotional quality of your vocal tracks. At one time or another, many of these tips have saved my bacon when technological and human elements conspired against me. I hope they'll serve you well.

Comfort Zones

Half the battle of capturing exquisite vocal performances is showing respect for the singer's craft. And this advice is even more important if you're the engineer, producer, *and* vocalist. Multitasking home recordists often don't cut their creative entities enough slack, opting instead to obsess on what they consider the more important technical details of a recording session. This is a big mistake. Sterling audio production is certainly critical, but a great recording of an uninspired performance still equals a bad track in my book.

Taking the time to prepare for a vocal session can make the difference

between a perfunctory reading and an impassioned performance. Whether you're working with a singer, or handling the vocal chores yourself, some simple groundwork can defuse any annoyances that can throw a vocalist off balance. Most of these tips are born from pure common sense, which makes it rather sad that many musicians refuse to get a clue and do what's right. In other words, don't allow your recordings to suffer because you're too lazy to create an environment that welcomes perfection.

GOLDEN SLUMBERS

Here's a big "duh" that musicians *still* decline to acknowledge: You can't sing like a bird if you're dog tired. Too many singers record their lead vocals after surviving a brutal eight hours on their day job, and they end up wondering why they can't generate any heat (or even sing consistently on pitch). Likewise, instrumentalists will slave for hours recording basic tracks and then decide to start singing. Bad idea.

For most of us, singing is not easy or automatic. To maximize your vocal gifts, you must ensure that mind and body are well rested. It's extremely difficult to concentrate on pitch, phrasing, and breath support—and be open to inspiration—when you're fatigued. So if you really want to sing your best, get some rest!

KEEP IT WARM

In the world of professional sports, pregame warm-ups are not just for show. Stretching exercises prepare an athlete's muscles for maximum exertion and protect against pulls, strains, and tears. Because the act of singing also involves delicate muscle groups, vocalists would do well to adopt a "presession" ritual. It's truly disconcerting how many singers will plunge into the act of recording without warming up their vocal chords. And walking into a session cold isn't just stupid, it's dangerous.

I worked with a rock singer who refused to warm up, and he ultimately spent a ton of studio time trying to hit notes at the upper tip of his range without his voice cracking. The vocal sessions could be more accurately described as an endless series of one- or two-note punch ins, and I noticed his range gradually degenerating as the project progressed. Soon, his speaking voice was also affected. A few weeks after the album was completed, a specialist found polyps on the singer's vocal chords. I don't think the recording sessions alone caused the polyps to form, but they certainly aggravated the problem.

The moral of this little tale is to keep your voice in shape, and the best way to pump up your vocal chords is to sing scales. Working out with a couple of major scales before singing full out can relax your larynx, focus your breathing (so you don't strain for air), and ensure that you hit notes clean and on pitch. I've found that singing scales also instills confidence, because your ears get acclimated to "hearing" pitch before you have to perform flawlessly for the tape.

Because running scales for as little as fifteen minutes can vastly improve

a vocal performance, it's hard to accept that so many singers simply blow off this critical exercise. So whenever you feel too lazy to do the prep work, just keep in mind that the tape doesn't lie; you may be listening to a substandard performance for a long, long time.

SAVE ME

Yet another mistake that recordists often make is not saving enough tracks for the vocal. Setting aside one track for the lead vocal just doesn't make it anymore. For one thing, you rob yourself of doubling a performance on separate tracks to increase sonic density. You also lose the chance at tracking an alternative—and possibly way better—performance once an acceptable lead vocal is cut. And you can forget about comping several different vocal tracks together to construct a killer take. Are you getting the idea that assigning one track for one vocal is downright dumb?

For pop songs composed of a single lead and a couple of background vocals, I recommend saving at least four tracks for vocals. This provides reasonable flexibility to comp those critical lead vocal tracks and stack up (or double) harmony and background vocals. Recordists who work on 8-track MDM systems should try submixing their rhythm instruments to two tracks, thus opening up six tracks for instrumental solos, sweetening, and vocals. (You could also mix your rhythm tracks to DAT and bounce them back to the multitrack as a stereo pair.) Believe me, once you remove the anxiety of having to cut a killer vocal onto one available track, the quality of your performances will soar.

TIME PASSAGES

Walking into a vocal session without a lyric sheet is tantamount to trekking through the Sahara without a map. Trust me, when the vocal overdubs become fast and furious, you *will* get lost. And once you're dazed and confused, the chances are darn good that you'll either erase a line you wanted to keep or forget to punch in some sections that needed improvement. Is avoiding a few minutes with your word processor worth all this added stress? I don't think so.

It's amazing how much a simple, double-spaced lyric sheet can support and guide a vocal session. It allows you to write the appropriate tape-counter numbers or absolute times next to each lyric line so you can roll right to a problematic phrase. (Now I know many recorders offer cue-point memory, but few have enough slots available for each and every line of a song.) In addition, it lets you follow the vocalist word for word as he or she sings, and you can use a pencil to circle the lines that are poorly articulated or off-pitch. When an entire performance is tracked, you can scan the lyric sheet for a definitive visual reference of troublesome lines. And you can confirm "fixes" by simply erasing the circles as each vocal miscue is repaired. There's truly no easier way to keep track of all the pearls and woe that fall from a singer's mouth.

MAKE ME SMILE

Creature comforts can play a large role in maximizing one's natural gifts. It's not exactly a trade secret that the more the vocalist can concentrate on singing, the better. I always make sure that a chair or stool is close at hand to allow the singer to sit down between takes and that a music stand is provided for lyric sheets. In addition, bottomless cups of herbal tea and mineral water should be placed within arm's reach. It's also important to do little things such as hanging the headphones on a separate stand for easy access, keeping the vocal booth (or recording area) free of claustrophobic clutter, and dimming the lights to calm nerves or enhance a mood.

On the more technical side, if a singer is having a difficult time nailing high notes, you can pitch down the tape speed to make things a little easier. This is like instantly transposing the song to a slightly lower key. Don't go nuts, however, because if you reduce the tape speed too drastically the vocal will sound like a chipmunk when the tape is played back at normal speed.

And speaking of pitch, many an engineer and producer have been driven mad by singers who can't hit the side of a barn, note-wise. But why make things so hard when you can cheat? Simply play the main melody and harmony lines on a piano and record them onto separate tracks. When the session begins, put the appropriate piano part into the headphone mix so the singer can use the instrument as a guide (or yes, a crutch). I haven't been marched to the brink of insanity by a singer for years, and it's all because this stupid little trick works like a charm.

The subtleties of microphone selection and placement can be as mysterious as voodoo rituals. I once did a textbook-perfect mic placement on a guitar amp and became very impressed with myself when I heard a glorious roar through the monitors. Well, the sound *was* glorious, but it had nothing to do with my engineering expertise. After I walked back into the control room, the oblivious guitarist had moved the mic stand three feet to the left of the speaker cabinet to make room for his stomp boxes.

There's a point to this embarrassing little story: textbook mic placement is just a starting point, you have to listen critically and be brave enough to move the mic around until it captures a divine timbre. You may be shocked at where the mic ends up—I once tracked a background vocalist by placing an AKG C414 two feet behind his head—but the quality of the sound you record is all that matters. This tenet is especially critical when recording vocals, because documenting the bountiful tonal colors and dynamics of the human voice is quite a chore. Be sure to explore all options before committing to a specific microphone and mic placement.

LISTEN TO THE MUSIC

Unless you've discovered the indisputable dream microphone for your (or your singer's) voice, don't begin a vocal session with just one mic. Each singer and song is different, which means one microphone can't possibly

enhance every type of performance with dazzling results. You need some diversity here, and there's nothing like a well-balanced mic menu to expand your audio options.

If it's a critical project—as opposed to a quickie demo—rent or borrow at least three diverse models: a large-diaphragm condenser (such as an AKG C 414, a Neumann TLM193, or an Audio-Technica AT4050/CM5); an affordable tube condenser (such as Peavey's PVM T9000 and the Groove Tubes' MD1); and a few medium- and large-diaphragm dynamics (such as the "standard" Shure SM58, Sennheiser's MD421, and Electro-Voice's RE20 and others).

Once you've assembled your mic collection, place them all on stands with the appropriate wind screens or pop shields and line them up in the vocal booth (or room). Run each microphone through a separate channel on your mixer and assign each channel to an individual track. Now have the vocalist sing an identical verse and/or chorus through each of the mics, recording each performance flat (without EQ adjustments). Make sure, for the sake of an easy audition, that you record these separate vocal tests at the same starting point. Obviously, the thing to do now is listen critically to each track and determine which microphone best enhances the singer's voice. I like to roll tape and mute and unmute each track to immediately compare the timbre of one mic to another within the same lyrical phrase. However, you can also learn a lot by listening to each of the short performances in their entirety and selecting the most evocative vocal sound.

If everything is set up and ready to go before the actual session, you usually can find the ideal mic in less than an half hour. That's a tiny bump in a production schedule—you probably spend more time labeling your tape boxes—and well worth the hassle. If you're committed to improving your vocal sound, abandon the single microphone theory.

IT'S NOT UNUSUAL

Okay, you *do* need a starting point, so I'll cough up a fundamental vocal miking scheme. Initially, most engineers tend to place the mic one foot from the singer's face, at his or her eye level. This position tends to favor upper midrange frequencies in the 3 kHz area for a crisp, well-articulated tone. It also avoids rushes of air that can cause pops and other unseemly distortions, as long as the vocalist can be encouraged *not* to raise his or her mouth towards the microphone.

POP MUSIC

While we're on the subject of unseemly distortions, I should emphasize that plosives and pops are devilishly hard to fix in the mix and should be dealt with at the recording session. The simplest solution is to move the mic farther away from the singer if he or she is consistently popping off. A windscreen or pop shield can also help, but keep in mind that foam models can absorb high frequencies.

If the singer is confident enough to take direction, try having him or her move his or her mouth slightly away from the mic when problem words such as "please" and "but" creep into the lyric. Of course, if you don't want to teach a seminar on mic technique in the studio, you can always move the mic off-axis to the singer's mouth. If all else fails, try cutting -10 dB in the area of 100 Hz with your console's EQ. This tonal tweak will not eradicate the problem, but it may diminish the boominess enough to make the track useable.

Sometimes you'll find that you just can't defeat a plosive. In these instances, you can either hope that the rhythm track covers the pop, or you can ask the songwriter to change the offending word to a less plosive one. Unfortunately, both of these options can be hazardous to your (mental) health.

HEART FULL OF SOUL

The late R&B legend Otis Redding is one of my favorite vocalists, but the stirring soul shouter must have been pure hell to record. You can actually hear him overload the mic on several of his classic tracks. (Aren't CD reissues wonderful for exposing the little boo-boos of the analog era?) Luckily, mic selection and placement can save contemporary engineers from being similarly victimized.

If the singer is shredding your condenser, consider using a less sensitive, large-diaphragm dynamic mic. (Remember how we tested all those different mics before the session?) The

Lip Service

overall response of the dynamic will not be as wide as the condenser's, so determine if the vocalist's range actually requires an expansive frequency spectrum. If not, you're home free. Slip a foam pop filter over the dynamic of your choice, position the mic approximately eight inches from his or her mouth, and dare the boisterous singer to overload a mic that can handle a snare drum hit without flinching.

Unfortunately, if you love the sound of the condenser, all you can do is move the mic further away from the mouth of doom. I'll often position the mic approximately two feet in front of the singer and hang it just over his or her head. This position captures a fair amount of room tone, but if the room sounds good, the combination of source signal and ambience can add sparkle to the track. Of course, if the reflections are bothersome, you can dampen them by hanging absorptive material wherever the ambience is the most obnoxious.

WHISPERS IN THE DARK

Airy, ethereal vocals can be intensely evocative. When Sinead O'Connor surrenders to a wounded whisper in "Nothing Compares 2 U," the effect is almost heartbreaking. Capturing the subtle nuances of a fragile voice is not easy, however, and it requires dogged collaboration between singer and engineer.

For one thing, recording a soft voice necessitates placing the mic dangerously close to the singer's mouth. Any plosives, coughs, or loud breaths will likely be fatal at such close range, especially when using a condenser mic. The vocalist should be made aware of the perilous situation and asked to respect the sensitivity of the microphone. For example, I often have a singer raise his or her hand if a sneeze is imminent to allow me to safely mute the input channel.

My favorite starting position for wispy vocals is a condenser mic placed approximately four inches from the singer's mouth, with the diaphragm pointing "dead on" his or her lips. About two inches in front of the mic, I place a wind screen to deflect breaths, and I ask the singer to place his or her nose right against the screen. Even if the singer has absolute control over his or her vocal dynamics, you still may have to adjust the mic's pad to -10 dB for overload insurance. In addition, try cutting 100 Hz on your console's tone controls by -7 dB or so to prevent excessive boominess.

This position really focuses on the nuances of the voice, especially if reflections are minimized by recording in a dry, carpeted room. Vocalists often find it difficult to control their pitch and dynamics when singing at low volumes, but they discover it's actually easier to emote when they're not trying to push large blasts of air through their larynx. But if the singer can maintain vocal control, you should be rewarded with an incredibly moving performance.

HARD TO HANDLE

So what can you do when a singer insists on holding the microphone in his or her hands to deliver an unrestrained performance? Well, sometimes the best mic position *is* hand-held. However, you don't have to toss the vocalist a mic and live with the handling noise and inevitable plosives.

First, you can try the Jim Morrison method. The sexy, wasted singer of The Doors often couldn't get psyched up to sing in the studio unless he could drape himself onto the mic stand and fondle the microphone (which is exactly the performance style he used for concerts). To ensure they got a track out of the erratic pop genius, the engineers would place a dead mic on a stand for Morrison to play with, and position the "real" mic a foot or so above his head. Pretty tricky, huh?

If the Morrison tactic fails, pop a heavy foam windscreen on a dynamic mic and wrap the mic's body in a sheath of one-inch thick foam. (It always pays to have various densities of foam available, so you can cut pieces to render services such as this.) Now

you can hand the mic to rowdy singers and let them bounce off the walls if they so desire. Handling noises and pops should be diminished, but there will be sonic compromises. You'll probably have to do some major EQ tweaks to keep the voice from sounding dull. (The protective foam absorbs high-frequencies.)

Also, because you can't control exactly where the singer will put his or her mouth during the performance, you may get obvious frequency shifts in the voice. For example, if the vocalist sings a verse right into the center of the mic, it will probably sound full and robust. However, if he or she invokes live performance techniques and pulls away from the mic for the song's louder chorus, the sound may thin out too much (depending on how far the mic is held away from the mouth). If the singer is really working the mic, the vocal tone may change quite dramatically within the same verse or chorus. However, you'll have to determine whether holding the mic has allowed the singer to cut a vocal of such power that the technical blemishes are reduced to near insignificance.

Not Fade Away

Recording vocals can be a stressful experience for both the singer and the engineer. Hopefully, these mic placement, mic selection, and studio "lifestyle" tips will help you face your next vocal session with a little less trepidation. And if this article has taught you anything worthwhile, it's that preproduction can ensure a smooth-running session.

The bottom line is: vocals are extremely important (and that's a major understatement). However, far too many musicians and recordists shortchange the process. I've watched countless demo sessions where the vocalist is expected to cut his or her tracks in one 4-hour session, after the band took days to record basics and overdubs. Small wonder that the projects often turn out disappointing. If your vocal tracks aren't happening, it may be time to readjust your priorities. ❦

Flash in the RAM

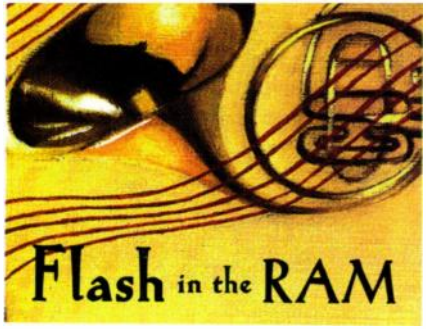
Does software-based synthesis mean the end for sound cards?

BY Paul D. Lehrman

When boat owners gather, one of the inevitable debates is whether inboard or outboard motors are better in terms of power, efficiency, and maintenance. Now the same argument is being heard in synthesis circles. Do we need external hardware or internal sound cards to produce music? Should we do it entirely in software?

With the new, super fast CPUs available for our favorite desktop platforms, there's a movement afoot to create software synthesizers. Thanks to chips such as the PowerPC and Pentium, as well as inexpensive digital-to-analog converters (DACs), companies are looking for ways to use the CPU to produce all-synthesized audio.

Illustration by Kitty Meeks



There are several good reasons for this approach. For one thing, it could provide a way to bypass the notorious difficulties of installing sound cards, particularly on DOS and Windows machines. IRQ conflicts, specialized driver software, competing synthesis technologies, and sound-library incompatibilities often cause nightmares when setting up and using a sound card. If these cards could be eliminated while preserving the sound quality, a big headache for the music-for-multimedia community would go away.

Of course, if you dumped your sound card, you would still need a DAC-converter in order to work with digital audio and route the synth's output to an external sound system. Macintoshes and a few PCs are already equipped with DACs that serve this purpose, but the vast majority of PCs lack this feature as standard equipment. This will be easily resolved if computer manufacturers build relatively inexpensive DACs into their machines, as expected. But in the meantime, don't throw away your sound card.

HISTORY

Software synthesis is not a new concept. Years ago, you could coerce a computer to play music by writing a machine-language program that triggered a speaker at speeds in the audio range. When you turned the speaker on and off 440 times a second, you got something approximating a square wave playing A.

In 1983, Commodore put a simple, 4-voice synthesizer chip known as a Sound Interface Device (SID) in its Model 64 computer, which was well used by game designers for the platform. Then, on the first Macintosh, Apple included a 4-voice, 8-bit, wavetable sound chip that could do real-time synthesis on the fly. It was supported by a number of software products that let you design sounds and play music; you could even use the sounds with a MIDI keyboard. Later, the AV Macintoshes included a 16-bit, high sample-rate sound chip, but most manufacturers used it for recording and playing digital audio files, not for synthesis.

Today's machines are so fast that dedicated sound chips are no longer necessary to perform real-time synthesis. The technology is straightforward. A bunch of audio samples that are organized in lookup tables called "wavetables" are loaded into the computer's RAM. When the operating system receives a command to play a note, the appropriate sample is retrieved, passed through a pitch shifter or other real-time processor, and sent to a DAC, which turns it into a sound we can hear. DACs cost about $50 or so at the retail level (considerably less at the manufacturing level), and they can be found on the motherboards of many "multimedia" computers, particularly slotless notebooks.

From the DAC, the sound proceeds as an ordinary analog signal to headphones or an amplifier and speakers. The commands controlling the process are normally MIDI messages sent through the computer's operating system using special drivers or extensions. Some examples of these drivers include Microsoft's Multimedia Control Interface (MCI) and Opcode's Open Music System (OMS), which has been licensed for use in future versions of both QuickTime and Windows.

PERFORMANCE

The quality of the sound depends on the usual parameters: sample resolution (8- or 16-bit) and sampling rate. However, another factor becomes important in software synthesis: the number of sounds that can be played simultaneously. The CPU spews out samples at a fixed rate, so the number of samples it can retrieve and play at one time is a function of its clock speed.

The polyphony is also affected by the amount of CPU power available for this task. (Don't forget, the CPU must often simultaneously handle such demanding tasks as video, graphics rendering, and/or digital audio.) A slow CPU may be able to generate four simultaneous, real-time "voices," whereas faster ones can handle more. If a system is touted as a true General MIDI (GM) synth, it must be able to play at least 24 voices; if there are fewer voices available, many scores will play incorrectly.

Yet another variable is the size of the sound set. A typical GM synthesizer has between 1 and 4 MB of sample ROM. In a software synthesizer, all of that data must fit into the computer's RAM.

Designers of software-synthesis systems can use some tricks to maximize performance without overwhelming the computer. To keep RAM requirements down, a system can "cache" sounds, loading only the sounds needed for a particular sequence just before playing it. Another technique is to compress the sounds, using JPEG or a similar algorithm, and decompress on the fly. This requires a coder/decoder, or *codec*. However, codecs and caching put an extra burden on the CPU. To minimize CPU load, sophisticated voice-stealing algorithms can limit the number of voices sounding at one time, without causing the music to suffer unduly.

There are nearly a dozen companies involved in software-based synthesis, designing synthesis engines and/or supplying sound sets. These products are being driven in large part by two major players: Apple and Intel. However, there are many within the music and multimedia communities who think the whole idea is overrated. They claim that software synthesis is not ready for prime time, and it may never be. The outcome of this debate will have a major effect on the future of music in multimedia.

MAC-BASED PRODUCTS

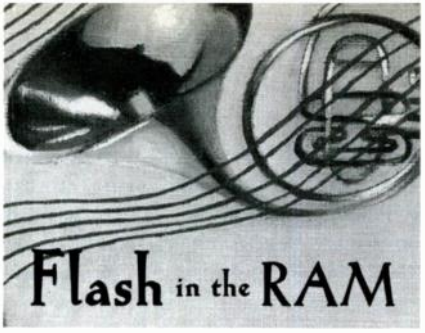
Apple's QuickTime 2.0 system extension includes a set of samples provided by Roland called QuickTime Musical Instruments. These samples were drawn from Roland's GM sound set and reduced to about 500 KB by judicious selection. A Standard MIDI File sequence can be added to any QuickTime movie. When the movie plays, the sequence plays the QuickTime Musical Instrument sounds.

The quality of the sounds depends on the Mac—8-bit for 68030 or '040 machines and 16-bit for AV and Power Macs—and the number of simultaneous voices ranges from four to six on a Mac LC to as many as 30 on the fastest Power Macs.

The first implementation of QuickTime Musical Instruments, released in early 1994, was pretty clunky. Only a few sounds from each GM instrument bank were present. When a Program Change called for a sound that wasn't there, it defaulted to the most appropriate substitution. However, if you imported a Standard MIDI File into QuickTime, program and bank numbers were shifted by one in the translation. Large SMFs tended to crash the system, and the only real-time synth controls were Velocity, Volume, and Pitch Bend.

Finally, there is a significant delay when playing the sounds: the software takes approximately 100 milliseconds to process a MIDI command and play a sound. If you're just playing back sequence files, this is no problem, because all sounds are delayed equally. However, synchronizing built-in sounds with external MIDI sound sources is difficult, and "playing" the QuickTime sounds from a MIDI keyboard is nearly impossible.

The next incarnation of the Musical Instruments extension, version 2.1, should be available by the end of the year, and it supposedly takes care of some of these problems. The Program Change/Bank Select bug is gone, and



large SMFs will no longer crash the system. More multisamples are included, and more real-time controllers, including Modulation, are recognized. "A lot of GM scores will play almost correctly," says an Apple spokesman.

The sound set is the same, but the system now allows third-party sounds to be included. Apple is reportedly looking closely at Sound Fonts, a proposed standard under development by E-mu, as a way to organize and exchange sounds. There's nothing in version 2.1 to address the delay problem, but my source at Apple says they are "going to make an aggressive effort to address it with the next go-round. We're looking to get it down in the 10 to 15 ms range."

Another Macintosh system has been announced by InVision Interactive of Palo Alto, California. *CyberSound GM* (see **Fig. 1**) is a subset of a high-end, professional synthesis system that will be available at a later date. The larger system will support multiple types of synthesis, including analog, PCM, physical modeling, and FM, complete with a complex modulation matrix, resonant filters that can be controlled in real time, and programmable time-based effects such as reverb and chorus.

The GM version will incorporate all of these technologies, but they won't be under direct user control. The system will load as a control panel, and it will be played using either MIDI Manager or OMS. "There's no delay problem," says spokesman Tim Gehrt. "We've been working on that specifically."

There will be two basic GM sound sets: one for 68000-series machines and one for the PowerPC. According to Gehrt, the PowerPC version will be comparable to any pro synth in terms of response time, but the 68000-series version will exhibit more response delay in favor of better performance on the slower CPU. Both sets use 16-bit samples. InVision has developed technologies such as dynamic sample-downloading and intelligent algorithm-scaling that manage the synth's load on the CPU, providing better overall performance. InVision expects to port the system to the Windows world "in the very near future."

WINDOWS-BASED PRODUCTS

Intel manufactures most of the CPU chips that are used to control MS-DOS and Windows computers. In addition,

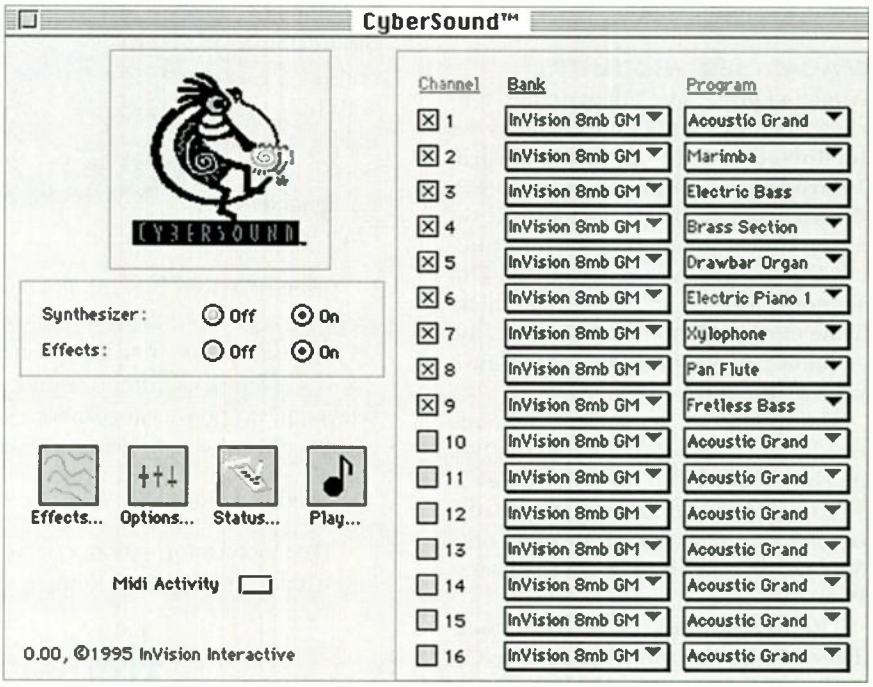


FIG. 1: *CyberSound GM* from InVision Interactive is a Mac-based software synthesizer. This screen lets you select your sounds and even includes a MIDI-activity indicator.

Over 100,000 served.

Yes, it's true. Since we introduced our MM-401 PC MIDI interface in 1991, MIDIMAN has sold over 100,000 MIDI interfaces worldwide. And all of them are *Guaranteed For Life*.

What's more, MIDIMAN offers the largest, most comprehensive and cost effective line of MIDI/Multimedia interfaces available.

Internal MIDI/Multimedia Interfaces for the PC—

Starting at only $69.95.

Winman–Available in either 1 in/1 out or 2 in/2 out models. Winman works with all software that supports the Windows Multimedia standard. Windows and Cakewalk DOS drivers included.

MM-401–The PC standard. 1 in/1 out MIDI interface. Fully MPU-401 compatible. Works with all DOS and Windows programs that support the MPU-401 standard.

External MIDI/Multimedia Interfaces for the PC—

Starting at only $39.95.

Portman–1 in/1 out models are available for both serial and parallel ports. 2 independent in/4 independent out model available for the parallel port. Perfect for laptop or desktop applications. Windows and Cakewalk DOS drivers and cable included with all models.

Sound Card MIDI Cable–Instantly (and inexpensively) connects your SoundBlaster™ or SoundBlaster™ compatible sound card to the world of external MIDI keyboards and sound modules.

Macintosh MIDI/Multimedia Interfaces—

Starting at only $40.00.

Mini Macman–World's smallest and least expensive MIDI interface. 1 in/1 out with MIDI in, out and power LED indicators. Compatible with all Macintosh MIDI software. Serial cable included.

Macman–1 in/3 out MIDI interface with MIDI in, out and power LED indicators. Serial thru switch allows you to select between printer (or modem) and MIDI. Compatible with all Macintosh MIDI software. Serial cable included.

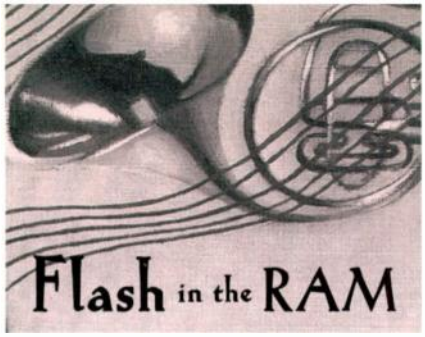
Mac Syncman–2 in/6 out MIDI interface, synchronizer and SMPTE regenerator. Professional performance priced for everyone. Supports all MIDI sync protocols including MIDI Time Code, Direct Time Lock and MIDI Song Pointer (Smart FSK). Full jam sync and flywheeling. Built-in studio quality SMPTE regenerator. Rack ears, Mac D/A and serial cables included.

MIDIMAN products are available at over 400 retail locations in the continental U.S. and in over 30 countries worldwide. You can also find many of our products around the world under such prestigious names as **Roland**, **Steinberg** and **EMAGIC**. For product information or the name of a MIDIMAN dealer or distributor near you, call us toll free at (800) 969-6434.

midiman®

GUARANTEED FOR LIFE

236 West Mountain Street • Suite 108 • Pasadena • CA 91103 • Tel: (818) 449-8838 FAX: (818) 449-9480 BBS: (818) 449-2019 • CompuServe: MIDIBVEN Forum(Section 2) -ID#75300,3434 for E-Mail



the company has had their own software-synthesis system, *SoftNotes*, up and running since last year. *SoftNotes* is the brainchild of Seer Systems, a Silicon Valley company started by Stanley Jungleib (formerly of the seminal synth manufacturer Sequential Circuits). Jungleib recruited his former boss and mentor Dave Smith as president of the company.

Intel will use *SoftNotes* as the basis of an upcoming product called *Native MIDI*. "The system can allocate 32 voices, unless there's something else going on with higher priority," says Jungleib. "The samples are 8-bit. There has been some talk of going to 16 bits, but we don't think it's really necessary. Even professional synths don't always take advantage of 16 bits."

Seer Systems' sounds and synth engine are also included with Turtle Beach's $99 Monte Carlo sound card. In this case, the engine is called *V-Synth* (see **Fig. 2**). According to spokesman Roy Smith, *V-Synth* works with any 80486 or Pentium system and any sound card "with the correct drivers." Right now, the Monte Carlo is the only available card that works with *V-Synth*. Smith expects the drivers in Windows 95 will make it possible to run the system through any Windows-compatible sound card. However, since the system depends on the virtual device architecture that Windows provides, it doesn't work with DOS games. Smith also hopes to have the company's *SampleVision* and *WavePatch* editing programs available for the new platform.

FIG. 2: Turtle Beach's *V-Synth* is one incarnation of Seer Systems' SoftNotes. Here, you can mix sounds from various sources.

Another Windows-oriented company is Brooktree. This 12-year-old San Diego semiconductor company has developed *WaveStream* (see **Fig. 3**), which works with Windows on 80486/33 and faster machines. The sound set, which includes 8 MB of 16-bit samples, is from Q-Up Arts, and the sample rate is fixed at 22 kHz. A caching system, which loads the required sounds with each sequence, uses between 600 KB and 2.3 MB, depending on the desired quality.

Real-time control is provided over Pitch Bend, Aftertouch, Mod Wheel, and Sustain, as well as dynamic filters, filter envelopes, and LFO. The number of voices is selectable on the fly using a feature called Dynamic Polyphony. If there is another FM card installed in the computer, a MultiSynth feature will use it to play some of the instruments, reserving wavetable voices for the most important parts. According to Joe Monastiero, senior product marketing manager for Brooktree, "With a '486/33, we recommend using nine wavetable voices, with the rest in FM."

Although the *WaveStream* software was developed with a SoundBlaster 16 sound card, Brooktree intends to ship it on several platforms, including their own MediaStream chip set. This hardware does audio, graphics, and video on a PCI card (see "Tech Page: Gently Down the MediaStream" in the January 1995 **EM**). Ports to other sound cards are expected shortly thereafter, and user-designed sounds will be accommodated through a program called *SampleXchange*.

IBM's Power Personal Systems division has announced *IBM SoftMIDI Synthesizer*, a software synthesis system for the company's PowerPC-based computers. It ships with the Windows NT 3.51 operating system and can play up to 32 notes simultaneously using 16-bit, 22 kHz samples. Various combinations of wavetable and subtractive synthesis can be used. A stereo, audio codec is included on the motherboard, along with a joystick/MIDI port that uses the same cable as a Sound Blaster.

Average CPU use is claimed to be

ART GIVES YOU A MUCH BETTER WAY TO WARM THE SOUND OF YOUR MICS!

If you've been getting a less-than-great, cold, flat sterile sound from your mics or instruments, it's time to warm them up. Get ART's new Tube MP Microphone Preamplifier. You'll suddenly find yourself with lush, rich, and yes, WARM sounding mics. The Tube MP utilizes a hand-selected 12AX7 to impart all of the warmth, and richness you want. Plus, it's got 70dB of low-noise gain, built-in phantom power, and a phase-reverse switch. Use it for digital (or analog) recording or live shows. It's great when used as a direct box (particularly on bass) or with transducer-style or piezo pickups. Best of all, it's inexpensive enough to put one on every mic you own.

TUBE-MP

- 70DB OF GAIN
- HAND-SELECTED 12AX7 TUBE
- BALANCED XLR & UNBALANCED 1/4" INPUTS
- BALANCED XLR & UNBALANCED 1/4" OUTPUTS

$149

- EQUIVALENT INPUT NOISE XLR-IN SPECS: -129DB
- DYNAMIC RANGE: >100 DB
- PHANTOM POWER TO 48V
- 20DB GAIN SWITCH
- PHASE-REVERSE SWITCH

SEE US AT AES Booth #744

A R T
APPLIED RESEARCH AND TECHNOLOGY

215 Tremont Street, Rochester, New York 14608 USA Phone 716-436-2720 • Fax 716-436-3942 • email: artroch@aol.com

For A.R.T.'s full color catalog (and to be put on our mailing list) please send $2.00, along with your name and address to: A.R.T. Catalog Dept., 215 Tremont Street, Rochester, NY 14608. Be sure to tell us what magazine you saw this ad in, and its cover date.

15% for most songs, and IBM plans to reduce this load in future versions. The system includes a software stereo mixer that can combine up to six mono or stereo samples with 8- or 16-bit resolution at rates of 11, 22.05, or 44.1 kHz. IBM reports that Sonic Foundry has ported its highly regarded *Sound Forge* 3.0 sample-editing software to run on the system in native PowerPC code.

The Kurzweil Technology Group is developing a software-synthesis engine for a number of large PC manufacturers. The sound set is almost identical to the 2 MB ROM that is included with several of Kurzweil's General MIDI chip sets, which they market to third-party manufacturers. The software features a maximum polyphony of 32 voices and a maximum sample rate of 44.1 kHz. According to third-party developer relations manager Fred Lapitino, "The number of voices will be software controllable, which allows the user to optimize the synthesizer structure for the varying demands of multimedia applications and PC hardware."

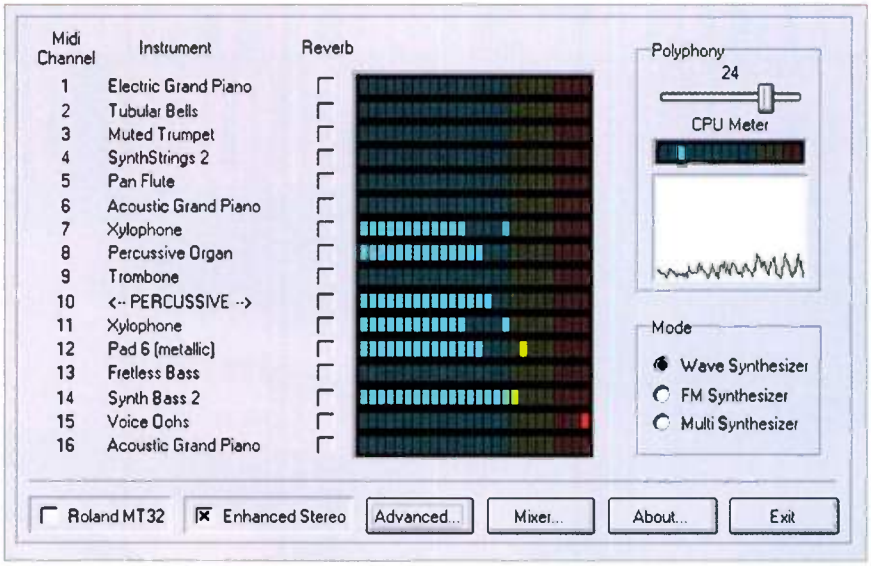
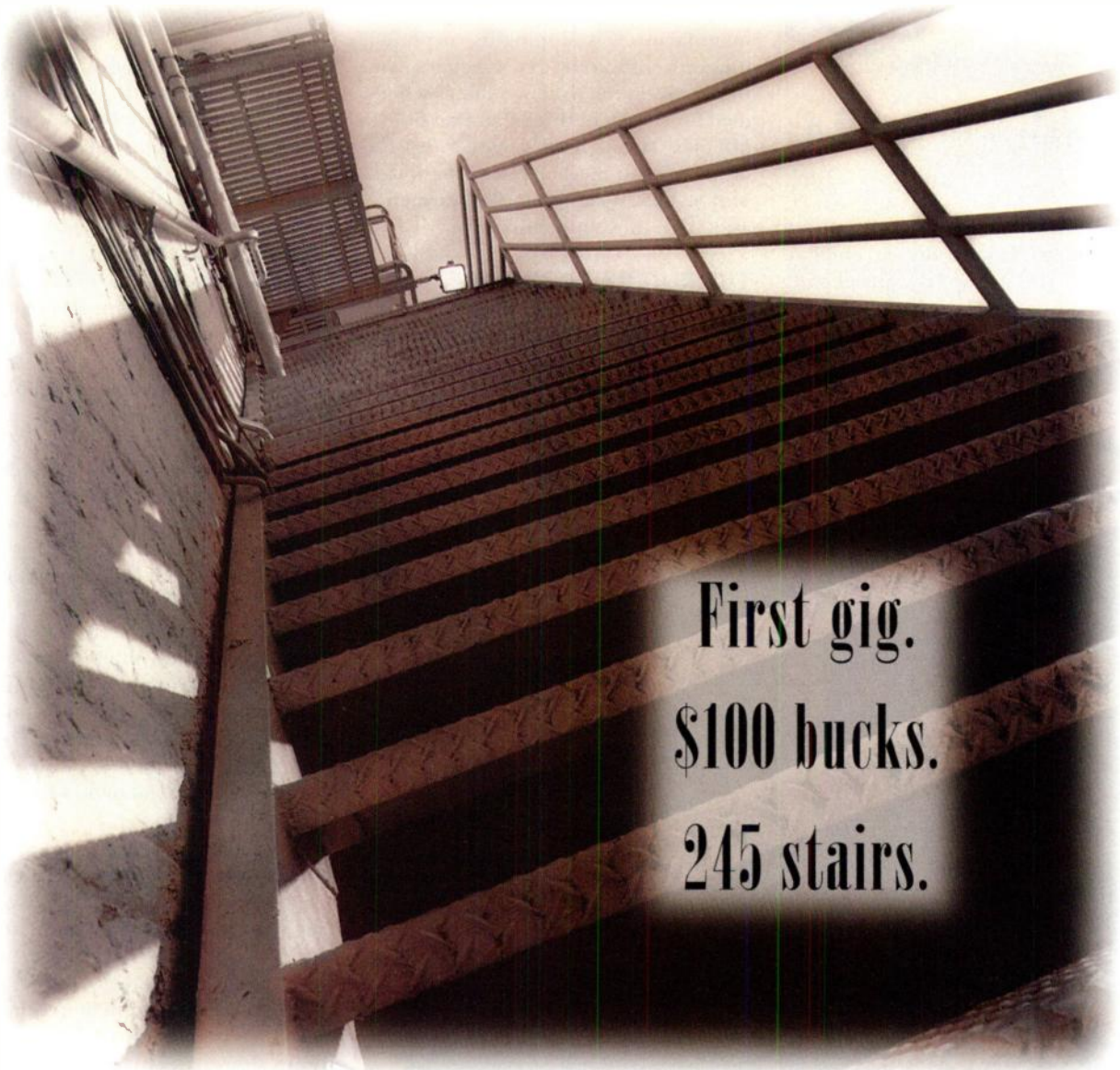


FIG. 3: Brooktree's *WaveStream* software offers some very attractive screens, including one that indicates individual instrument levels and CPU loading.

The synth can be loaded with new samples using the MIDI Sample Dump Standard. However, a physical MIDI cable isn't involved, so the operation is quite fast. The size of the sound set is not fixed; it can expand to use all the



SOMETIMES THE DIFFERENCE between taking a gig and not taking it isn't the money. Or the exposure. But the schlep. Which is why we created the new EON Portable System. At about half the weight of any other system, it's made to be portable. Yet it gives you all the sound quality and reliability that JBL is known for. At a price that is noticeably lighter. In fact, it may be the perfect first step for any rising star who's just starting out. And plans to get to the top.

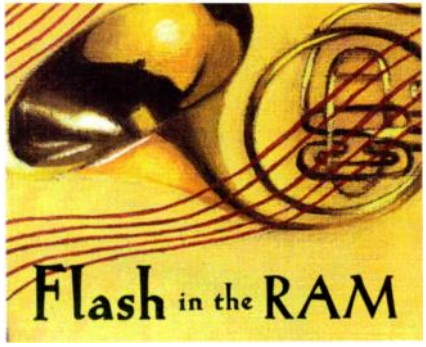
EON™

JBL

H A Harman International Company

S O U N D T H A T C A R R I E S

See us at AES Booth #418



RAM available in the host computer. According to Howard Brown, chief architect of the group, the processing delay for live MIDI control is "under one millisecond, except that the operating system sometimes gets in the way."

Altec Lansing, a well-known speaker manufacturer, is dipping its corporate toes into software with a wavetable synthesis system for Windows that is scheduled for shipment this fall. It uses only 1 MB of RAM, and it will run on a 75 MHz or faster Pentium. The price is minimal, too: $29.95. It can even play wavetable and MIDI files simultaneously.

THE END OF HARDWARE?

With so many major companies backing software synthesis, is this the inexorable wave of the future? Is hardware doomed, at least as far as PC-based music is concerned?

In truth, it's still too early to throw out your sound cards. As mentioned earlier, most PCs lack DACs, which are required for routing the synth's output to a sound system and for working with digital audio. In addition, the performance "hit" on a computer—even a fast one—that is trying to generate 32 voices in real time can be substantial, and some developers feel that playing music is not the most efficient use of a CPU's time.

According to Kurzweil's Lapitino, "Playing 32 high-quality voices at 44.1 kHz uses up almost all of the average '486 and about half the processing overhead of a 100 MHz Pentium." His colleague Brown agrees: "Software synthesis is a great advertisement for hardware synthesis. The hit is considerable. At a sample rate of 44.1 kHz, you're looking at 2 to 5% per voice on a 90 MHz Pentium. It's a false economy; it looks like the synth is free, but it's actually taking up as much as 60% of your CPU. When you add in the cost of the RAM that is dedicated to doing music, you're spending $750 to do what a $150 sound card can do."

Some might consider Brown's figures to be a bit high, but many in the audio community share his concern. Gordon Currie, an audio designer at Microsoft, says software synthesis is "a great idea, but it still has a ways to go. It's the same as other software implementations of stuff that originated in hardware: it's always slower in software and not as good quality."

"Everybody and his brother is developing software," Kurzweil's Brown continues, "assuming there's so much idle time to spare with the new CPUs. However, if *all* your DSP is in software, you end up with a dog-slow processor that's trying to do everything at once: audio decompression, video, 3-D graphics, and a modem in the background. The hardware is finally fast enough that we can start writing code to match, but if everybody does it in software, the system will slow down again. The user will think, 'This is ridiculous; I paid all this money, and it's all hype. The system is slow as molasses!'"

Others point out that hardware-based synthesis is cheap, so it makes more sense to use the CPU for tasks that can't be easily duplicated in hardware. "A 120 MHz Pentium with 16 MB of

RAM has enough memory and processing power to do high-quality synthesis," says Microsoft's Geoff Dahl, "but some games would prefer to use the resources for improved 3-D rendering or video and audio decompression. The computer may be able to do one or two of these tasks at once, but it probably wouldn't do a great job with all four at once."

SOFTWARE SYNTHESIZER MANUFACTURERS

Altec Lansing tel. (800) 648-6663 or (717) 296-2818; fax (717) 296-1222; e-mail 7536.351@compuserve.com

Apple Computer tel. (408) 996-1010; WWW http://quicktime.apple.com

Brooktree Corp. tel. (619) 452-7580; fax (619) 452-1249; e-mail apps@brooktree.com; WWW http://www.brooktree.com

E-mu Systems tel. (408) 438-1921; fax (408) 438-8612; WWW http://www.creaf.com

IBM Personal Computing Company tel. (800) 472-7693; fax (800) IBM-4FAX; WWW http://www.austin.ibm.com/powrinfo.html

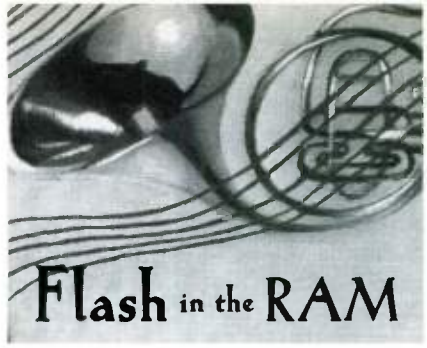
Intel Corp. tel. (800) 538-3373; e-mail nsp@intel.com; WWW http://www.intel.com; ftp ftp.intel.com/pub/ial

InVision Interactive tel. (415) 812-7380; fax (415) 812-7386; e-mail dennym@cybersound.com; WWW http://www.cybersound.com

Seer Systems tel. (707) 963-7006; fax (707) 963-2183; e-mail daveseer@ix.netcom.com

Turtle Beach Systems tel. (717) 767-0200; fax (717) 767-6033; e-mail sales@tbeach.com

Young Chang R&D Institute tel. (617) 890-2929; fax (617) 890-2014; e-mail ktg@ycrdi.com; WWW http://www.musicpro.com/kurzweil/



Flash in the RAM

Dave Rossum, who is director of the E-mu/Creative Labs technology center, calls software synthesis "a cool thing to use your computer for. But right now, anything that sounds interesting takes a significant hit of computer power. If you only want to synthesize music, that seems to work. It's not necessarily the best sound quality or most accurate sequence playback, but I don't retch and vomit when I hear it. However, any reasonable synthesis takes 25% or more of a CPU's performance, so you must sacrifice something when you get into processor-intensive applications."

On the other hand, even the skeptics see a role for software synthesis in the future. "Music professionals will never accept the performance of software synths," says Turtle Beach's Smith, "but for games, it can be quite happening. As time goes on, the proportion of processor power will diminish by orders of magnitude until it gets to the point where it's a background task, like a screen saver."

Rossum agrees. "In the long term, as the machines get faster by an order of magnitude, we will drop from 25% to 2.5% of the processor's time. Then, synthesis becomes small change. Of course, we continue to develop new things that may keep demand for audio performance above CPU performance. For example, our latest work involves multichannel 3-D audio, which is computationally intensive. In the future, however, people will be very happy if they can get the equivalent quality of a 1995 GM synth from software."

So who will win the software synthesis game? Perhaps Fred Lapitino has the right philosophy. "The people who have been doing it the longest will do it the best. All wavetable software synths basically do the same thing, but it's the craft of the people putting the sounds together that will make the difference."

Paul D. Lehrman *is the coauthor of* MIDI for the Professional, *available from Mix Bookshelf, and is the director of the Pro School program at the University of Massachusetts, Lowell.*

You'll Sound Better Tonight

(Guaranteed or your money back!)

Purchase a SoundTech system. That's all you have to do, regardless of size or number of components. From a basic solo performance PA to a full touring rig, if you purchase an entire SoundTech system, we guarantee your sound will improve. We're so confident of our sound quality that, if for any reason you're not fully satisfied, you'll get your money back. Tear out this ad and present it to your SoundTech dealer and tell him you want the SoundTech **"Sound Better Tonight"** guarantee.

(If you think this deal sounds good, wait until you hear our systems)

(Bring to your SoundTech dealer)

3 YEAR BUILT TOUGH WARRANTY

It's Simple, You'll Sound Better!

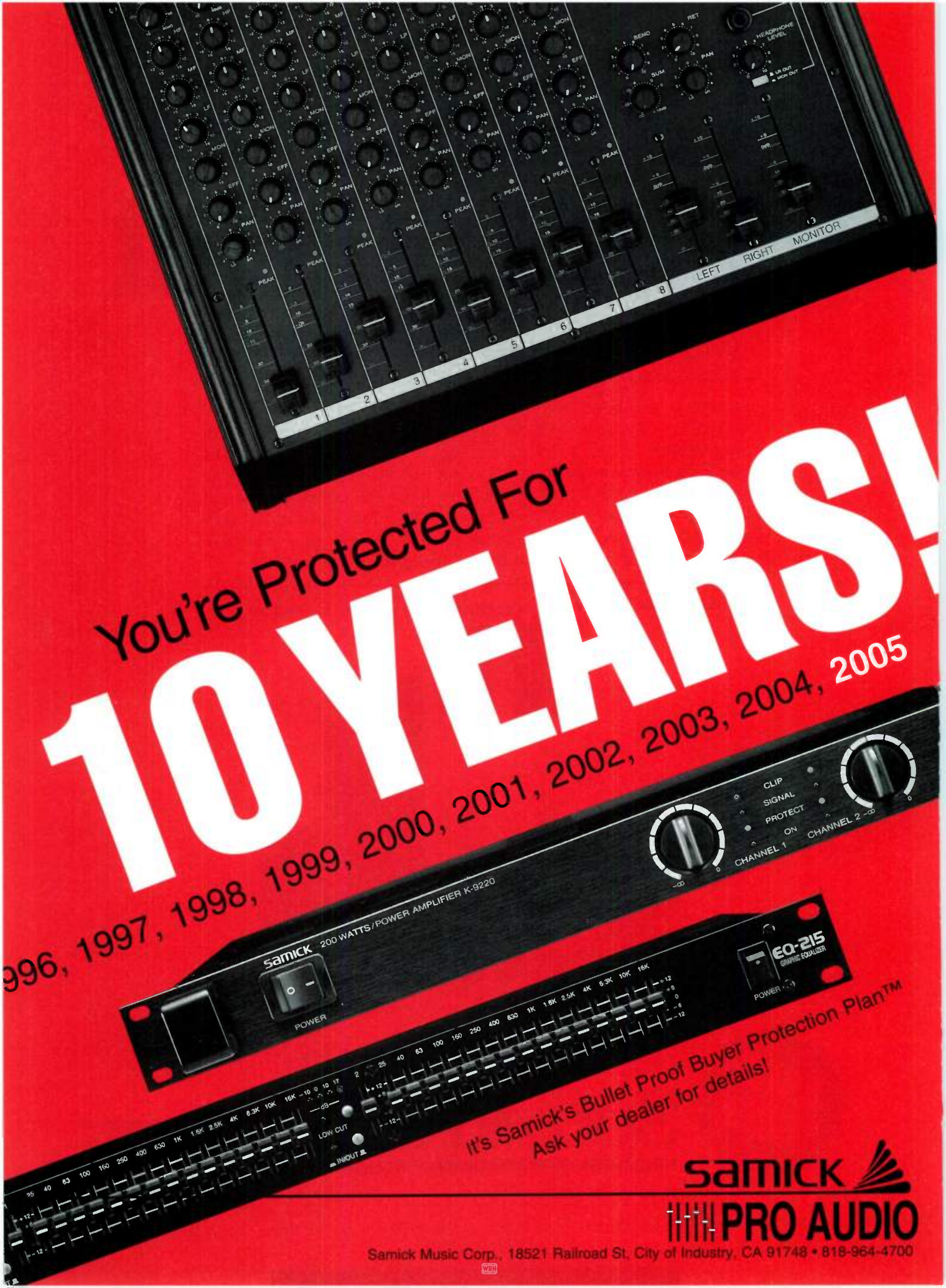
SoundTech™

PROFESSIONAL AUDIO

255 Corporate Woods Parkway • Vernon Hills, IL 60061 USA
(708) 913-5511

For the location of the participating Sound Tech dealer nearest you... Call 1-800-US-SOUND, ext. 131 / 132

Complete **"Sound Better Tonight"** details available at participating dealers or from SoundTech.


You're Protected For
10YEARS!
996, 1997, 1998, 1999, 2000, 2001, 2002, 2003, 2004, 2005
Samick 200 WATTS/POWER AMPLIFIER K-9220
POWER
CLIP
SIGNAL
PROTECT
ON
CHANNEL 1
CHANNEL 2
EQ-215
GRAPHIC EQUALIZER
POWER
LEFT
RIGHT
MONITOR
It's Samick's Bullet Proof Buyer Protection Plan™
Ask your dealer for details!
samick
PRO AUDIO
Samick Music Corp., 18521 Railroad St, City of Industry, CA 91748 • 818-964-4700

MULTIMEDIA MUSICIAN

Going Full Throttle

LucasArts revs up digital sound for their latest CD-ROM game.

By Peter McConnell

Digital audio is no big deal to most **EM** readers. Many of you are old hands on hard-disk recorders, modular digital multitracks, and DAT machines. However, complete digital audio soundtracks are still a rarity in the interactive computer-game industry. This situation has been a drag for game-sound pros such as myself, who have yearned to use sound effects, professional vocals, and high-quality music production in a way that approaches the sonic experience of movies and television.

Recently, we at LucasArts Entertainment were finally able to emulate the cinema experience for the soundtrack for our CD-ROM *Full Throttle*, a PC-based motorcycle adventure game. We wanted to intensify the game experience by using digital recordings of real actors, real motorcycles, and a real hard-rock band. But it was no joyride bringing these elements together, and we had to do an elaborate dance between art and technology to produce LucasArts' first all-digital, fully interactive game soundtrack.

It's a long, hard ride for tough guy Ben in LucasArts' new CD-ROM game *Full Throttle*.

KICK STARTING

Full Throttle is an adult cartoon in which the player is Ben, the larger-than-life leader of a motorcycle gang. Ben lives in the post-apocalypse American West and has been framed for murdering the founder of the last remaining motorcycle company. The real killer, Adrian Ripburger (voiced by Mark Hamill of *Star Wars* fame), plans to turn the proud motorcycle firm into a minivan manufacturer. He also happens to want Ben dead. Of course, it's up to Ben to clear his name and stop Ripburger's insidious plot, while trying not to get killed.

Scoring this harrowing adventure was a challenge. We needed to evoke two distinct emotional pictures: the loneliness and desolation of the landscape that is the backdrop for the game's puzzle-solving scenes and the roaring bravado of bikers on a rampage for the action sequences.

If this doesn't change your tune, it will add a whole lot of new ones to your repertoire.

QS300 Introducing the world's first XG synthesizer and a powerful music production workstation. Complete voice editing for ultimate control. 947 XG, GM, preset and user voices for maximum expression and compatibility. 3,093 editable musical phrases with "play effects" for infinite creativity (over 1,000 drum and percussion tracks, 575 bass lines, 500 guitar chords and riffs, and 800 keyboard chords and riffs). Three totally programmable effects processors with realtime control for total production capability. And a huge screen and elegant GUI interface for ease of use. Go from inspiration to ovation without ever leaving the QS300.

XG

XG

YAMAHA®

For more information on the QS300, call (800) 932-0001 Ext. 610. ©1995 Yamaha Corporation of America, Digital Musical Instrument Dept., P.O. Box 6600, Buena Park, CA 90622

Creatively speaking, it would have been difficult to capture the true essence of these sonic vistas if we were limited to the sounds generated by sound cards. Happily, the digital audio playback from the CD-ROM allowed us the flexibility to choose any sound sources we wanted. For the music during the problem-solving scenes, for example, I was able to compose ambient orchestral pieces with strong hints of Ry Cooder and Ennio Morricone. Deadline pressures prevented me from using actual acoustic instruments for these underscores—it would have been near impossible to find the right players, rehearse the parts, and schedule studio sessions within the production time line—but I was able to get some great sounds from my Kurzweil K-2000, E-mu Proteus, and Roland Sound Canvas.

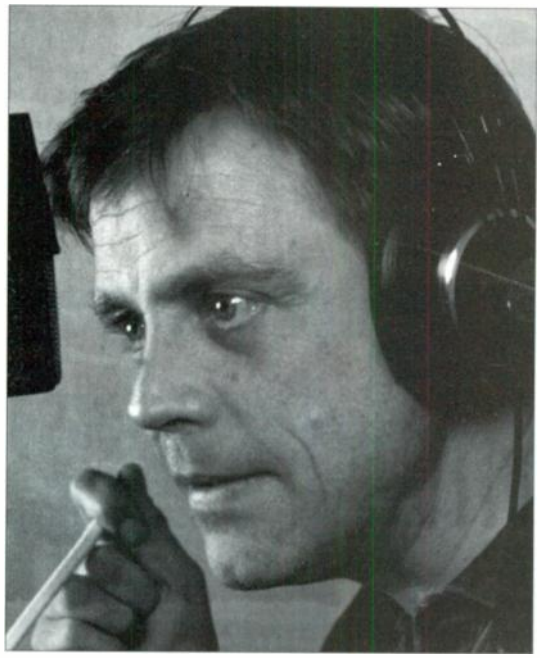
Because *Full Throttle* could play back recorded sound, we also had the option of using a live band for the action-oriented scenes. This is where the availability of digital audio really enhanced the game. After all, there is only one way to evoke the visceral power of roaring bikes, and that's with pounding, hell-bent-for-leather rock 'n' roll. After some frustrating talks with "name" recording acts, we found a treasure in our own backyard: the Gone Jackals. As a bona fide biker band, the Jackals possessed the right attitude and sound to bring the essential hard-rock integrity to *Full Throttle*. Even better, the band already had a full CD's worth of material recorded and ready for us to use.

Having a digital audio soundtrack in *Full Throttle* allowed us to do more than create smashing music tracks, it also let us produce feature film-quality sound design. To create the roars and rumbles of futuristic motorcycles, for example, Sound Designer Clint Bajakian often mixed a bunch of different sounds together—which is exactly how the audio wizards at Skywalker Ranch produced many of the spacecraft sounds for *Star Wars*. Bajakian spent countless hours in the field recording motorcycles and then countless more hours in the studio manipulating the sounds. One trick he used was combining animal sounds with engine noises to make the bikes seem alive and menacing. (Try to pick out the sound of a growling Bengal tiger when Ben's motorcycle is idling.) Other bike sounds were married with recordings of donkeys, lions, and even bees buzzing in a jar.

INTERACTIVE SCORING

However, we couldn't pat ourselves on the back and say "job well done" just yet. The real challenge of scoring *Full Throttle* was ensuring that all the underscores and the sound design blended seamlessly into the game's interactive environment.

Unlike conventional film and television productions, computer games do not follow linear chronologies. Interactive media is, well, *interactive*. You can't predict what a player will do (or when he or she will do it), and yet the soundtrack must follow the game play without a single glitch. How well the score accommodates the demands of interactivity is largely determined by the game's sound-playback technology. If sounds must keep pace with unpredictable actions, the playback system must be capable of accessing several sounds simultaneously to ensure that the right sound is ready to go when the gamer makes his or her move. In addition, if music tracks are expected to



***Star Wars* alumnus Mark Hamill provided the voice for the game's evil Adrian Ripburger.**

change as events unfold, the system must also be able to perform smooth transitions between different pieces of music.

LucasArts developed its proprietary iMUSE (interactive music and sound effects) system in 1991 to address these problems for MIDI scores played through PC sound cards. Unfortunately, directing the playback of digital music is a tougher nut to crack. For one thing, the music is no longer a relatively tiny, 20K Standard MIDI File loaded into memory. Digital audio tracks take up many megabytes of data, all of which must be pulled off the CD-ROM as the game is being played. Such mammoth processing demands called for a simple playback system to ensure seamless operation.

Therefore, the iMUSE system for *Full Throttle* was designed to perform easy crossfades between music tracks and to play up to six digital sounds at once, which allowed us to mix speech, music, and sound effects to deliver a rich sonic experience. For example, when Ben explores the town of Melonweed, it isn't always clear whether you are hearing sound effects or music. Because iMUSE is capable of playing multiple "tracks," we could constantly bring the sounds of animals, junkyard machinery, and the music underscore in and out of the mix as Ben moves around his environment.

Another challenge of interactive scoring is maintaining sonic cohesiveness between thousands of sound files. In an interactive game, you can't always predict which sounds will be heard along with something else as the player navigates through each scene—there are just too many possible combinations to audition. However, Ben's voice must always be articulate and clearly audible, and whatever constitutes "loud" music must play back at roughly the same volume level throughout the game.

The snag is that there is currently no way to hear all the files as they would actually play in a scene, nor can you make real-time signal-processing tweaks. If you want to make an effect louder and brighter, you must grab the desired sound off a disk, run separate passes of software EQ and gain processes, run a conversion utility to lower the sample rate, and repress the corrected audio onto a single CD-ROM. This process can take more than an hour or two to do what would amount to one slider move and one knob twist on a conventional mixer/recorder combination.

To maintain a relative sonic balance between the thousands of sound files in *Full Throttle*, we developed ways to play some files against each other without having to burn them into the game. We also tried to audition sounds we knew were likely to be heard together. However, we still ended up making a lot of guesses and compromises.

BANDWIDTH BUMMERS

Even the basic sound quality of the game was far from a done deal. Despite the pristine nature of digital resolution, we had to jump through some hoops to produce dazzling audio for *Full Throttle*. The reason for our sonic gymnastics is that current technology would not allow us to stream digital audio from a CD-ROM at the pro standard of 44.1 kHz, 16-bit stereo without compromising graphics performance.

The Gone Jackals' wild rock score is hard enough to melt the ink from the tattoos of real live bikers.

To ensure that *Full Throttle*'s animation responded swiftly to game play, we were limited to 22 kHz, 8-bit monaural sound. (The mono elements were ultimately panned in a stereo field.) Therefore, any audio processing we did had to take into account that the final result would have no frequency components above 11 kHz. Also, very quiet sounds had to be avoided because the audio quality of low-level signals degrades badly at 8 bits.

The bit-bashing process is best exemplified by what we did to make the Gone Jackals' music work within the CD-ROM's audio constraints. We had to remix all of the songs in mono and make versions with and without vocals. Because the Jackals' master tapes were in several formats—2-inch, 24-track analog; 1-inch, 16-track analog; and 16 tracks of ADAT—we used three Bay Area studios for the remix sessions. You should have seen the reaction of the engineers when we told them the mixes had to be mono, have nothing audible above 11 kHz, and yet still sound full and balanced! In the end, we had to compensate for the high-frequency rolloff by cutting the low end to ensure the tracks wouldn't sound too boomy.

After the remixes were completed, we still had some tweaking to do. We put the music through several stages of digital editing and dynamics and EQ processing, using Quadra 650s armed with Digidesign's Sound Designer II and various software plug-ins. It was critical that the Jackals' rock cuts and the ambient music sounded tonally similar so that they could segue into each other.

Sometimes, we had to "get nasty" to get good sound. For example, the rock music actually sounded best when the signal gain was pumped up beyond digital clipping before we downsampled the tracks to 22 kHz. Some of these files were boosted 35 percent *after* normalizing! Given the tools available at the time, however, I believe that all of our improvising and nonstandard audio practices produced a sizzling and evocative soundtrack. If we had followed all the rules, I'm sure we would have sacrificed the game's sonic punch.

GEARING DOWN

The *Full Throttle* soundtrack exemplifies the challenges that music producers and sound designers must face to push the audio envelope of computer games. Although we still have a way to go before computer sound rivals that of films and television, the creative freedom offered by digital audio should inspire interactive artists to produce exciting and innovative CD-ROM scores.

Peter McConnell *composes music and designs music software for LucasArts Entertainment. When he's not doing that, he's writing songs, singing, and playing electric violin in the Bay Area band Dervish.*

WORKING MUSICIAN

Networking with E-Mail

The Web isn't the only place musicians are making connections.

By Jennifer Seidel

Public consciousness has been awakened to the Internet, and music magazines are raving about the ways that it can enrich musicians' lives and careers. But to really exploit the online explosion, you should already have a home page on the World Wide Web. You *do* have a Web page, don't you? Oh well, if Web pages are the next destination on the superhighway, a lot of us must still be creeping along the on-ramps.

However, e-mail accounts are much more accessible than Web pages and can still offer musicians easy access to thousands of potential fans. Internet mailing lists are stampless, stapleless, photocopyless versions of "hard copy" mailing lists. But this is not cyber junk mail! Addressees actually look forward to getting e-mail because they *subscribe* to the list.

The good news for musicians is that you can use these online lists to publicize tour dates, announce CD releases, provide ordering information for your record, or familiarize fans with your work. And the access that Internet mailing lists have to predefined markets is unparalleled. The tools to promote and market an entire career are online and available to any ambitious and savvy artist.

DMITRY PANICH

MEET YOUR FUTURE FANS

Although many mailing lists frown on shameless self-promotion, they *do* encourage performers to talk about their projects. For example, Folk_Music's welcome message says: "The connections that we have helped create between artists and their music have won us the attention of national record labels and management firms....If you represent an artist or venue that we should be hearing about, tune us in, will ya?"

One way to turn a mailing list into a promotional tool is simply to introduce yourself to the other subscribers. Ellis Paul, a well-known singer/songwriter, attributes a lot of his popularity to Folk_Music. "The folk list has been important in creating a national buzz

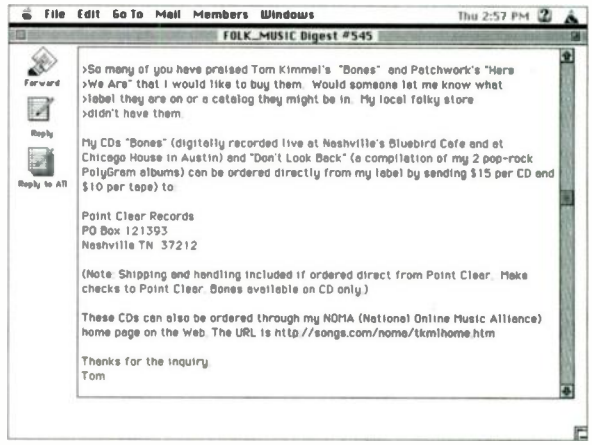
about my music," he says. "It's like a grapevine on steroids that's big enough to stretch across the U.S. I rarely play a show where there are no Folk_Music enthusiasts."

Once people know your name, they may want to hear your music or see you play. "Conventional mailing lists are only composed of fans who sign up at shows," says Paul. "But through the folk list, I can reach thousands of people who have never heard or seen me. These people subscribe because they are interested in the genre and then may hear other folk fans talking about me on the list. Sometimes the dialog will result in someone buying my record or driving out to see me play. It's a built-in marketing demographic because the folk list is full of folk fans. I can't miss!"

Some of the greatest mailing-list success stories are about musicians who have had shows, or even entire tours, booked through their connections on the Internet. One such story concerns folk duo Pete and Maura Kennedy, who posted the schedule for their first tour on some mailing lists.

"It was a good itinerary for a first tour, but there were a lot of holes in it," they explain. "The next day, our e-mail box was flooded with messages from coffeehouse promoters across the country who saw our schedule and offered to help fill in the holes. We ended up scheduling about a third of our tour from these mailing list contacts."

Some mailing lists even specialize in concert listings. A few of these lists are purely regional, while others are genre based. And smaller mailing lists often send their information to Musi-Cal, a national music-calendar database that lists shows by artist and city. (Musi-Cal can create custom calendars for each user and even send weekly updates if requested.) In addition, mailing-list subscribers often make each other aware of new or little-known venues when they post reviews about shows they've seen.



Here's an example of a fan and a musician finding each other on a listserv. The fan had heard great things about the artist on the list and wanted to buy his CD. The artist responded to the post himself.

Michael Hagen, a New York City publicist and booking agent for the nonprofit Fast Folk Cafe, suggests saving all your pertinent mailing list postings on floppy disks or on your hard drive and using them as an informal database. Then, you can use your word processor

SIGN ME UP

Here's a small selection of the many music-related mailing lists. A description of the list is given when it isn't made obvious by the list's name.

The text in brackets under each e-mail address is the subscription message to be typed in the body of your e-mail. Where the message reads "yourname," simply type your name. For example, to subscribe to the bluegrass list I would type *sub bgrass-l jennifer seidel*

Note: Each list will have its own rules about promotion. The lists shown here do not necessarily allow subscribers to post self-promoting messages.

Acid-Jazz
listserv@ucsd.edu
[subscribe acid-jazz yourname]

Ambient Music
ambient-digest-request@hyperreal.com
[subscribe]

Bluegrass and Old-Time Country Music (BGRASS-L)
listserv@ukcc.uky.edu
[sub bgrass-l yourname]

Blues Music (BLUES-L)
listserv@brownvm.brown.edu
[sub blues-l yourname]

Celtic Music (Celtic Houston Digest)
majordomo@chron.com
[subscribe celtic-houston-digest]

Classical Music
listserv@brownvm.brown.edu
[sub classm-l yourname]

Country
maiser@rmgate.pop.indiana.edu
[send subscribe country-l]

Film Music
listserv@jubvm.ucs.indiana.edu
[subscribe filmus-l yourname]

FOLKDJ-L
(for folk and bluegrass deejays)
listserv@psuvm.psu.edu
[subscribe folkdj-l yourname]

FOLKVENU
listserv@psuvm.psu.edu
[subscribe folkvenu yourname]

Gospel
maiser@rmgate.pop.indiana.edu
[subscribe gospel-l]

Hardrock
listproc@lists.colorado.edu
[subscribe hardrock yourname]

Medieval & Renaissance Music
mailbase@mailbase.ac.uk
[join med-and-ren-music yourname]

Midwest Raves
listserv@csd.uwm.edu
[subscribe mw-raves yourname]

Musi-Cal
concerts@automatrix.com
[help]

New American Folk Music
listserv@nysernet.org
[subscribe folk_music yourname]

North East Raves
listserv@umdd.umd.edu
[sub ne-raves yourname]

Orchestralist
listproc@hubcap.clemson.edu
[subscribe orchestralist yourname]

Progressive Music
majordomo@darktow.gun.de
[subscribe prog-request yourname]

Punk/Hardcore
punk-list-request@cpac.washington.edu (inside North and South America)
(subject should read: add)

San Francisco Raves
sfraves-request@hyperreal.com
[subscribe]

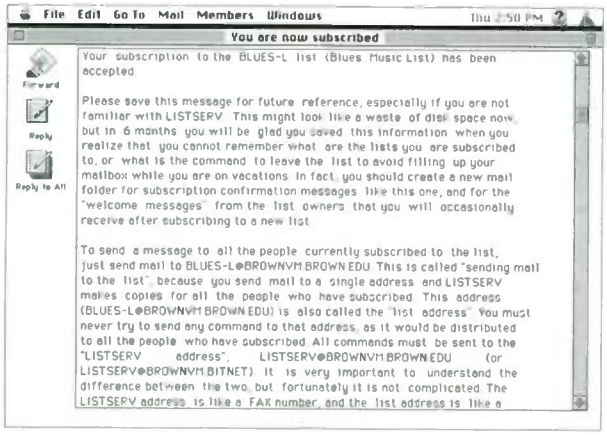
SYNTH-L
listserv@american.edu
[subscribe synth-l yourname]

to search the documents for key words or names.

"My collection of posts serves as an up-to-the-instant resource to find an e-mail address or phone number," he says. "I can find out which label or booking agency handles an artist or get the exact title of a new release. And artists can collect a few months' worth of postings and have a rich database of venues, festivals, agents, deejays, stations, labels, etc., that no print media can match."

If you can't be bothered compiling all this valuable information yourself, lists such as FOLKVENU act as virtural storehouses of industry data. The folk-oriented FOLKVENU, managed by Steve Jerrett, describes itself as "an opportunity to compare notes on such topics as bookings, promotions, finances, contracts, and general operations. Presenters also can share their schedules for the season and post short reviews of shows, while artists and managers can use the list to aid in scheduling."

Mailing lists are also a good place to find out about regional and national music festivals. Organizers will often post information about applying for festival slots or even ask fans to send booking suggestions. Sometimes organizers will approach artists that they've seen discussed on mailing lists directly. Also, musicians often alert each other when they hear about upcoming music-industry showcases—great places to meet other musicians and industry representatives.



When you subscribe to a mailing list, you should receive a welcome message that describes the list's intentions, general etiquette, and other "administrivia."

PROMOTE SALES AND AIRPLAY

As important as industry information can be to the independent artist, mailing lists can be much more than community "callboards." The opportunities for commerce are quite exciting. For example, Chris McMahon of the band Millan & Kenzie has used mailing lists to boost CD sales.

"We've sold hundreds of CDs and tapes all over the country, often to people who've only ever heard of us through Internet mailing lists and Web pages," he says. "Before we even went on the road, we had sold records in Seattle, San Francisco, Ohio, New York City, Boston, and countless other places because people heard about us on the Internet."

Of course, successfully hawking your CDs and tapes takes more effort than simply posting a message about their availability. You are going to need radio airplay, too, and mailing lists can put you in direct contact with deejays. Deejays collect in various places on the Internet, posting their playlists and eavesdropping on fans' conversations. FOLKDJ-L, for example, is a listserv for folk and bluegrass deejays that is managed by Tina Hay.

Singer/songwriter Nancy Moran has made the most of the access she has to deejays via FOLKDJ-L. "If I feel that my music fits a deejay's format, then I introduce myself via e-mail and see whether he or she would be interested in a copy of my CD," she says. "So far, everyone I have e-mailed has accepted, and I have seen my CD included on subsequent playlists. The deejays on the list seem to be incredibly receptive. Geoffrey Huys, a deejay at WGCS in Goshen, Indiana, told me that a musician he knew went from being played on ten stations to more than 100, just by e-mailing the list!"

You don't need a CD to benefit from these deejay contacts, either. John Wright was able to use FOLKDJ-L to find deejays to play his cassette release.

SHOPPING FOR MAILING LISTS

Here are two compilations of mailing lists. The first is for music-related lists. If you send an e-mail to the address with the message given here in brackets, you will then receive a copy of the master list. You can also access the list on the World Wide Web.

The second is a master list for all types of mailing lists, but the provided address will take you right to its music directory.

List of Musical Mailing Lists
E-mail address: avalon-request@dfw.net
[send lomml]
WWW address: http://server.berkeley.edu/~ayukawa/lomml.html

Publicly Accessible Mailing Lists
WWW address: http://www.neosoft.com/internet/paml/bysubj.html#music

"I don't have to waste time and money sending cassettes to stations who won't play them," he says. "I can just e-mail the deejays and ask whether they ever play cassettes. Naturally, it's particularly gratifying when a cut from my cassette later appears on one of their playlists."

JOIN THE CLUB

Don't forget that the Internet is a vehicle to meet people—real, live, breathing human beings—and not just their computers. Increasing your fan base, getting better attendance at your shows, selling your music, and having your songs played on the air are all results of making contacts with fans, venue owners, booking agents, and deejays. And as with any human interaction, there are rules of etiquette that need to be observed.

Because the amount of self-promotion allowed by different lists varies, make an effort to find out what is allowed before you post anything. A subtle, but effective way to sneak in a plug about your recordings or shows is to mention them in your signature file. Other subscribers read your .sig every time you post on the list, so they can contact you if they're interested in what you're doing—and you don't have to invade the list with self-accolades.

"The most important thing to remember is that people are sick to death of hype," counsels Alan Rowoth, Folk_Music's moderator. "Every artist's press kit has material designed to make him or her look like the biggest thing since Elvis. The net result is that everything directly posted by these artists and their support people is taken with a *huge* grain of salt."

And here's one final "golden" rule: Don't post on a list that you haven't subscribed to first. It's disrespectful. List administrators can spot uninformed posters a mile away and will probably yank your post before it gets disseminated to other subscribers. Join in the conversation, study the dialog, but don't post and run. Your plugs, when they do appear, will be better received when members recognize you as one of the gang.

Maybe by the time she records a demo, **EM** *Editorial Assistant* **Jennifer Seidel** *will have discovered all the great ways to promote it. Until then, she can be reached at emeditorial@pan.com.*

Space: The Final Frontier

3-D audio processors boldly go where no sound has gone before.

By Scott Wilkinson

The holodecks in the 24th century universe of *Star Trek* represent the ultimate simulation. Within the black-and-yellow confines of a relatively small room, the computer recreates any environment you care to program. The boundaries of this environment can appear to extend far beyond the walls of the room via holographic projection.

Although holodecks are currently restricted to the realm of fiction, it is possible to experience some of their magic in the audio domain with the help of 3-D audio processors. These devices improve conventional stereo playback by extending the apparent *soundstage* (the area from which the sounds seem to come) beyond the physical location of the two speakers. Many of these processors can also affect each individual source of sound (instrument, voice, etc.) in the recording, making it appear to emanate from a specific point around the listener.

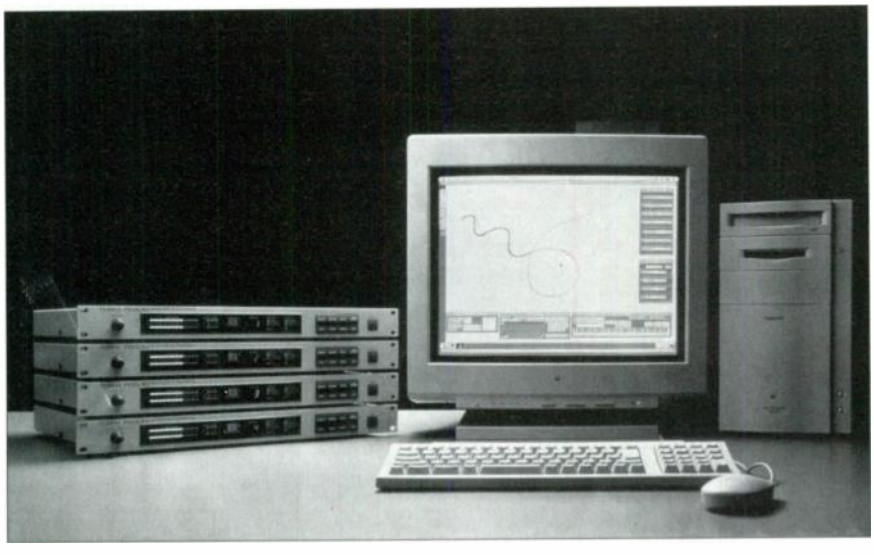


FIG. 1: Roland's RSS-10 system includes hardware and software to process stereo signals.

NATURAL LOCALIZATION

Before we look at these processors, though, we must understand how humans determine the direction and distance from which a sound is coming; this is called *localization*. There are four basic aural cues that help the brain localize a sound source: amplitude, timing, timbre, and ambience.

The effect of amplitude on localization is easy to see. Sounds from nearby sources are louder than those from sources that are farther away. In addition, if a sound comes from the right, it is louder in the right ear. Timing is also straightforward: if a sound comes from the left, it reaches the left ear first. This delay can be as short as a few microseconds, which amounts to a phase shift from one ear to the other.

The effect of timbre on localization is not so simple. For one thing, low frequencies travel farther than high frequencies. This is why elephants can communicate over great distances using subsonic vocalizations. As a result of this effect, distant sounds appear to be

THE ULTIMATE MUSIC RETAILER

PLUS HUNDREDS OF OTHER PRODUCTS FROM GENELEC • AKG • JBL
TANNOY • LEXICON • SUMMIT • OPCODE • FOSTEX • AKAI
MARK OF THE UNICORN • KAT • APHEX • ART • DBX • FATAR
DIGITECH • SHURE • TOA • SOUNDCRAFT • TWELVE TONE
SENNHEISER • TAC/AMEK • CAD AND DOZENS MORE!

THE BEST SELECTION OF THE FINEST PRODUCTS FROM OVER 100 OF THE BEST NAMES IN THE MUSIC BUSINESS...

That's right, we have the hottest new equipment in stock right now in our giant warehouse, ready to ship right to your door. What's more, our friendly, knowledgeable sales engineers make sure what you're buying is exactly what you need. And we back up each and every sale with the best technical support and after-the-sale service in the industry. Plus, with our exclusive "no hassles" money back guarantee, you simply can't go wrong. So what are you waiting for? Pick up the phone and call us and see why over 73,000 musicians call Sweetwater first for all their hardware and software needs!

VISA MasterCard DISCOVER

OUR EXCLUSIVE GUARANTEE:
"IF YOU DON'T LIKE IT, WE'LL TAKE IT BACK — WITH NO HASSLES!"

219•432•8176

5335 BASS ROAD, FT. WAYNE, IN 46808

FAX: 219•432•1758 / sales@sweetwater.com
http://www.sweetwater.com

Sweetwater SOUND INC.

ARE YOU KIDDING? Of course I want to receive a full year of Sweetwater Sound's **SWEET NOTES** newsletter absolutely free!

NAME_____
ADDRESS_____
CITY_____ STATE_____ ZIP_____
PHONE ( ) _____

Mail coupon to address above or call for your free subscription

1

Sonic Boom

"(Sound Forge) consistently performed in a rock-solid, glitch-free manner."
- Jim Aikin, Keyboard, May 1995

"Sound Forge succeeds in making itself the first truly complete professional sound editing system for Windows."
- Macromedia User Journal, March 1995

"This is what great software is all about."
- Electronic Musician, May 1995

Sound Forge Blasts the 16 Bit Barrier

Introducing Sound Forge for 32 Bit Windows™ – digital sound editing for Windows, Windows NT™, and Windows® 95 operating systems. Now all the features users have valued in Sound Forge are also available in a powerful new 32 Bit Windows version. Sound Forge for 32 Bit Windows is a full-featured sound editor intended for power hungry musicians, sound designers and multimedia developers seeking the performance benefits provided by high end 32 Bit Windows operating systems. This true 32 Bit flat model version of Sound Forge provides increased performance along with all the features available in the 16 Bit version. Both versions are unrivaled in power, making Sound Forge the most comprehensive digital sound editor for Windows.

"Sound Forge is a cool and full-featured program that's definitely a useful addition to any digi-head's toolbox"
- EQ, April 1995

"Sound Forge is so well written, so well documented, so powerful...it nearly brought tears to my eyes!"
- Seth Ritter, The Computer Program

For more information call Sonic Foundry at 1 800 57 SONIC (577 6642)

S O N I C  F O U N D R Y

100 South Baldwin, Suite 204, Madison, WI 53703, Tel: (608) 256 3133 Fax: (608) 256 7300, CompuServe: 74774,1340 or GO SONIC, Internet:sales@sfoundry.com

rolled off in the high end. In addition, as a sound reaches you, it is diffracted, reflected, and absorbed by your head and shoulders, which affects the timbre. As the sound enters your ears, it is affected by the acoustical properties of the *pinnae* (the large outer ear flaps) and the ear canal that leads to the ear drum. The exact timbral effect depends on the direction from which the sound comes.

Whenever a sound is made in an ambient environment, you hear both direct and reflected sound waves. Assuming there are no intervening reflective surfaces, the closer you are to the source, the higher the proportion of direct sound to reflected sound you hear. The arrival times of the reflections depend on the position of the sound source, which provides additional localization cues.

SIMULATED LOCALIZATION

Unfortunately, the recording and playback process distorts or removes many of these subtle localization cues. Microphones do not respond to sounds from different directions in the same way ears do; they have no head or pinna to affect the sound. In addition, stereo speakers do not reproduce the effect of localized sounds very well, especially outside the physical location of the speakers. You'd need many speakers arrayed around the listener to achieve this effect.

As a result, many companies have undertaken extensive research to determine exactly how we localize sound. They have then used this research to develop various ways to simulate localization by applying digital signal processing (DSP) to a stereo signal. The result is a number of products for the recording engineer and end user.

Although these products are commonly called "3-D audio processors," this is a bit of a misnomer. They can simulate an expanded soundstage from right to left rather effectively, but it is extremely difficult to simulate up-down placement with a single pair of speakers. Nevertheless, 3-D processors can be useful in a number of situations, including musical recordings, video and film soundtracks, multimedia, and computer games.

Most 3-D audio systems describe the position of a sound source in terms of distance, azimuth, and elevation. Azimuth is expressed in angular degrees around the listener's head. An azimuth

3-D PROCESSOR MANUFACTURERS

Crystal River Engineering
tel. (800) 317-TRON or (415) 323-8155;
fax (415) 323-8157

Dolby Laboratories
tel. (415) 558-0200; fax (415) 863-1373

QSound Labs
tel. (403) 291-2492; fax (403) 250-1521

Roland Corporation US
tel. (213) 685-5141; fax (213) 722-0911

RSP Technologies
tel. (810) 853-3055; fax (810) 853-5937

Spatializer Audio Laboratories
tel. (818) 227-3370; fax (818) 227-9750

SRS Labs
tel. (714) 442-1070; fax (714) 852-1099

of 0° is directly in front of the listener, whereas 180° is directly behind. It is very difficult to simulate a sound source behind the listener with stereo playback, so most 2-speaker systems expand the soundstage to encompass an arc of 180° around the front of the listener.

As you might expect, elevation is expressed in degrees above the listener. An elevation of 0° is directly ahead of the listener, while 90° is directly overhead. In some cases, you can specify a sound source's static position or program a moving source's trajectory and speed. However, few 3-D processors can effectively simulate elevated sound sources with stereo playback.

Simulating amplitude cues is easy: simply make the sound louder in one channel. Timing cues are also relatively easy to simulate: delay each sound by different, minute amounts in each speaker. As mentioned earlier, this delay is so short that it is often accomplished by phase shifting.

Timbre cues are more difficult to simulate. Researchers have developed mathematical models of the timbral effect of the head, shoulders, and pinnae, which are called *head-related transfer functions* (HRTFs). These HRTFs are implemented with powerful DSP. Equalization also plays an important role, especially dynamic EQ, which changes according to the input signal.

All speaker-based systems suffer from the same problem: the sounds from the speakers are acoustically combined in the air between them and the listener. In addition to hearing each speaker's sound in the appropriate ear (right speaker in the right ear, left speaker in the left ear), you also hear the right speaker's sound in the left ear and vice versa. This makes it even more difficult to simulate localization effectively.

One solution to this problem is to use headphones, which isolate the left and right channels from each other. An effective technique in this regard is called *binaural recording*. Tiny microphones are mounted in anatomically correct ear models on a dummy head, which is placed in the vicinity of a sound source (e.g., in the audience at a concert). The signals from the mics are affected by the shape of the dummy head and pinnae as they are routed directly to the right and left tracks of a stereo recorder. Then, these tracks are played back on headphones without interacting in any way. The HRTF effects of the dummy head are captured automatically, providing a convincing soundstage. However, wearing headphones all the time is impractical, so the search for speaker-based 3-D audio continues.

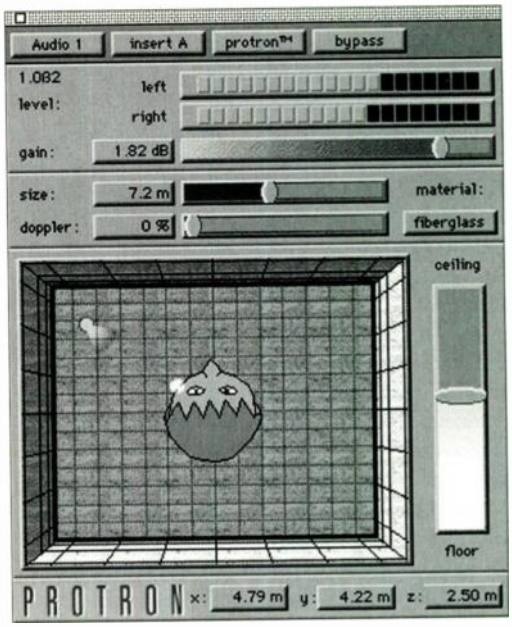


FIG. 3: Crystal River Engineering's *Protron* plug-in lets you specify the position of a sound source in three dimensions.

TWO-SPEAKER SYSTEMS

As mentioned earlier, most 3-D audio processors use standard stereo playback with two speakers. In addition, virtually all of these systems are *single-ended,* which means they are applied only at one end of the recording/playback chain. Most are applied to a mixed stereo signal as it is mastered for duplication, but in some cases, you can apply 3-D processing to several individual sounds as they are mixed. Finally, there are several systems that are applied to a conventional stereo signal during playback. (For a more detailed description of these systems, see "3-D Audio" in the October 1992 **EM**.)

One of the major players in this field is Roland. The Roland Sound Space (RSS) system is one of the few that claims to simulate full spherical localization, including right-left, up-down, and front-back. Introduced in 1991, its first incarnation was an expensive processor available only to the top of the recording profession. Since then, it has been repackaged in a less-expensive, single-rackspace unit, the RSS-10 (see **Fig. 1**), with control software for Windows or the Macintosh. The RSS process is also available in the SDE-330 Dimensional Space Delay, SRV-330 Dimensional Space Reverb, and SDX-330 Dimensional Expander. The process is designed to be applied during mixdown and requires no decoding during playback.

The RSS-10 accepts two input signals (right and left) and sends each one through two processors. The Binaural processors apply the HRTF that corresponds to the desired position of the sound source. This includes amplitude, timing, and timbre adjustments. The Transaural processors cancel the "crosstalk" signals (i.e., right speaker to left ear and vice versa). The effect is most pronounced if the listener is centered between the two speakers at the "sweet spot."

Hughes took a different approach when they developed the Sound Retrieval System (SRS) to enhance in-flight audio. They eventually offered a consumer-oriented box that enhanced the soundstage during playback of conventional stereo signals. Since then, a separate company called SRS Labs was started to develop the technology and license it to third-party manufacturers.

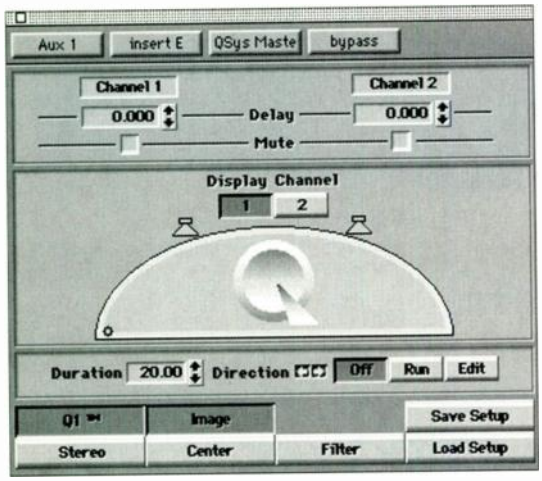


FIG. 2: QSound's *QSys/TDM* plug-in lets you position a sound source within a 180° soundstage. In this example, the source is in the extreme left corner of the soundstage.

(SRS is now found in TVs and other consumer-electronics products from Sony, RCA, and Packard Bell.) Multimedia companies such as MediaVision, NuReality, and Genoa Systems license SRS for use in sound cards, games, and other products.

The basic idea behind SRS is to restore the localization cues that are lost in the recording and playback process. SRS separates the sum (L+R) and difference (L-R and R-L) components of the stereo signal and applies corrective transfer functions, called *selective emphasis* and *de-emphasis*, to each component. This restores the ambience that is otherwise masked by louder direct sounds. In addition, it widens the soundstage while localizing individual sounds. SRS can also synthesize a stereo signal from a mono input. The company claims there is no sweet spot, unlike many other systems.

Spatializer Audio Laboratories' Spatializer is a single-ended 3-D processor that can be used at either end of the audio chain (during recording or playback). The process affects the difference signal by boosting the frequencies that provide the most important spatial cues (300 to 3,000 Hz). The difference signal is also slightly delayed to produce a sense of spaciousness.

The Spatializer is available in several forms. The original unit accepts up to eight signals (expandable to 24), each of which can be placed and moved around in the expanded soundstage. The Digital Spatializer accepts a stereo input and produces a stereo output. There is also a chip that manufacturers can include in their products. Licensees include Matsushista (Panasonic), Sharp, Hitachi, Ad Lib, and Orchid.

QSound is a single-ended 3-D process that is normally applied during the recording/mixing stage. QSound was first available only in the QSystem, an expensive unit that processes up to eight inputs using HRTFs derived from over a half-million listening tests. Many artists, such as Sting, Madonna, and Pink Floyd, have applied QSound to their recordings. In addition, several TV soundtracks, such as *Picket Fences*, *Wonder Years*, and *X-Files*, use the process. Multimedia and computer licensees include Atari, Sega, Analog Devices, Sound Source Unlimited, IBM, and Intel.

QSound Labs now offers several products in addition to the QSystem. For example, QXpander is a stereo process based on the QSystem's algorithm. This stereo process is available in several forms, including a family of chips that should find their way into inexpensive hardware products. The company also offers two software plug-ins for the Digidesign environment: *QSys/TDM* (see **Fig. 2**) and *QXpander for Sound Designer II* (see "The Hitchhiker's Guide to Plug-ins" in the July 1995 **EM**).

Crystal River Engineering also offers 3-D processing in both hardware and software products. Their approach is called AudioReality, which refers to an HRTF-based, real-time, immersive, 3-D audio rendering technology that localizes each sound independently as it is mixed. The company's first product was the Convolvotron, a PC-based system developed in conjunction with NASA/Ames Research Center for flight simulators and other virtual-reality applications (see "Tech Page: Virtual Reality Audio, Part 2" in the August 1993 **EM**).

Since then, they have created the Acoustetron II stand-alone system, the Beachtron and Alphatron PC cards, and a Digidesign Pro Tools plug-in called *Protron* (see **Fig. 3**).

MULTISPEAKER SYSTEMS

In the end, it's very difficult to coax clear localization out of two speakers. As a result, the film industry converged on multispeaker setups for commercial cinemas and, later, for home-theater systems. This approach doesn't require HRTFs or other complex sonic manipulations, but it is not single-ended. The signals destined for each speaker must be encoded during mastering and decoded during playback.

Dolby Laboratories is the most prominent name in this field. The Dolby Surround Pro•Logic system is designed for soundtrack applications (see "From the Top: Dolby Surround" in the April 1994 **EM**). The process encodes two extra channels—center and surround—in the main stereo program. Dolby doesn't sell the encoder; you must lease it when you mix.

However, the decoder is widely available in consumer receivers, TVs, and surround processors. This decoder extracts the center and surround signals and sends all four channels to the appropriate speakers: front left and right, center, and rear left and right. (Both rear speakers get the same surround signal, which is rolled off below 100 Hz and above 7 kHz.) If the program is played on a conventional stereo system, it sounds normal.

The next generation of the company's multichannel audio is called Dolby Surround AC-3. In this scheme, all five speakers receive their own independent, full-bandwidth signal. There is also a separate, low-frequency effects channel. These audio channels are digital, not analog. However, the audio is compressed to reduce storage and transmission requirements, a process to which some purists object. Currently, few products include an AC-3 decoder, and few laserdisc titles include an AC-3 soundtrack, but you can expect to hear more of this format in the near future.

RSP is another company that offers a multispeaker surround system. Called Circle Surround, this system is designed specifically for musical applications. The system includes an encoder (which you can buy), a controller with four joysticks, DOS-based control and automation software, and a decoder. Four separate inputs are encoded into a stereo signal, which is then decoded into as many as eight discrete, full-bandwidth channels. The company claims that Circle Surround is fully compatible with Dolby Pro•Logic decoders. In addition, the decoder can synthesize center and surround channels from any conventional stereo signal.

Although holodecks are a super cool idea, they remain beyond our current technical capabilities. However, it is possible to simulate various aural experiences from the comfort of your favorite chair with 3-D audio processors. These systems take one small step for their manufacturers and one giant leap beyond the aging stereo format without rendering it obsolete, offering a new dimension that should open a lot of ears to new possibilities.

EM *Tech Editor* **Scott Wilkinson** *enjoys listening to CDs, video programs, and movies that have been encoded with surround channels.*

Go To Your Room

Q2 2.0

Actually, with the powerful reverb algorithms in the Alesis Q2™, you can go to any room you want.

Q2 has the power to take your music to shimmering halls, bright chambers, dark caverns, or anywhere your creativity leads you. There's 300 carefully-designed reverbs, delays, EQ, pitch and special effects that provide the high-end professional processing you need without breaking the bank.

But the best reason to use a Q2 is the incredible flexibility it gives you to create your own space. Because the place where the music sounds the best – the room to be in – might be the room you call your own.

"I love the Q2. I'm using it in the show every night."
— Robert Scovill, 3 Time TEC Award Winner (Sound Reinforcement Engineer)

"I love that Q2. We're using the hell out of it."
— Ray Benson (Asleep At The Wheel), Multiple Grammy® Winner

"Q2 is the presetter's fantasy and the tweaker's dream."
— Francis Buckley, Top Independent Dance/Pop Engineer

- Stereo or Independent Dual-Channel Operation
- 100 Preset And 200 User-Editable Programs
- Octal Processing™—Up to Eight Effects At Once, In Any Order
- 24-Bit Internal Processing; 18-Bit D/A Converters; 48kHz Sampling Rate
- +4dBu Balanced and -10dBV Unbalanced Operation
- ADAT® Optical Digital I/O
- Custom LCD Graphic User Interface with Virtual Patch Cables
- **NEW!** Version 2.0 Software Now Features Up To 5 Seconds Of Sampling, Plus Overdrive, Surround Encoding, Triggered Pan with Doppler and More!

See us at AES Booth #556

Lezlie, Pitch Shift, 2-Band Sweep, 3&4-Band ... separate L/R input levels ... For more Q2 info, see ... trademark of NARAS.

ALESIS
STUDIO ELECTRONICS

Alesis Corporation 3630 Holdrege Avenue Los Angeles CA 90016 310-841-2272 alecorp@alesis1.usa.com

40-INPUT PROBLEM SOLVER.

PERFECT FOR KEYBOARDS!

Not every musician starts out needing a line mixer with 16 STEREO channels. But if you're serious about keyboards, sequencing, digital multitrack recording or electronic drumming, you'll be surprised at just how fast you'll grow into a Mackie Designs LM-3204.

Of course, when you consider everything that we packed into the LM-3204, it's a great mixer to start out with, too. You get two studio-quality mic preamplifiers, virtually every feature of our famous CR-1604 (and then some), plus twice the number of line inputs. All for under $1000 suggested retail*.

Same low noise/high headroom gain structure as the CR-1604. Same bottleneck-free mix amp architecture. Same musical-sounding 3-band EQ. And the built-like-a-tank construction quality that's made our mixers legendary on world tours and in 24-hour-a-day production facilities.

Then we added a Control Room output with its own 45mm fader, Tape Monitor section with both RCA and 1/4" inputs and outputs and even –20dB signal present LEDs on every channel. Then we made the LM-3204 expandable — you can plug in one or more LM-3204Es for 32 or even 48 inputs.

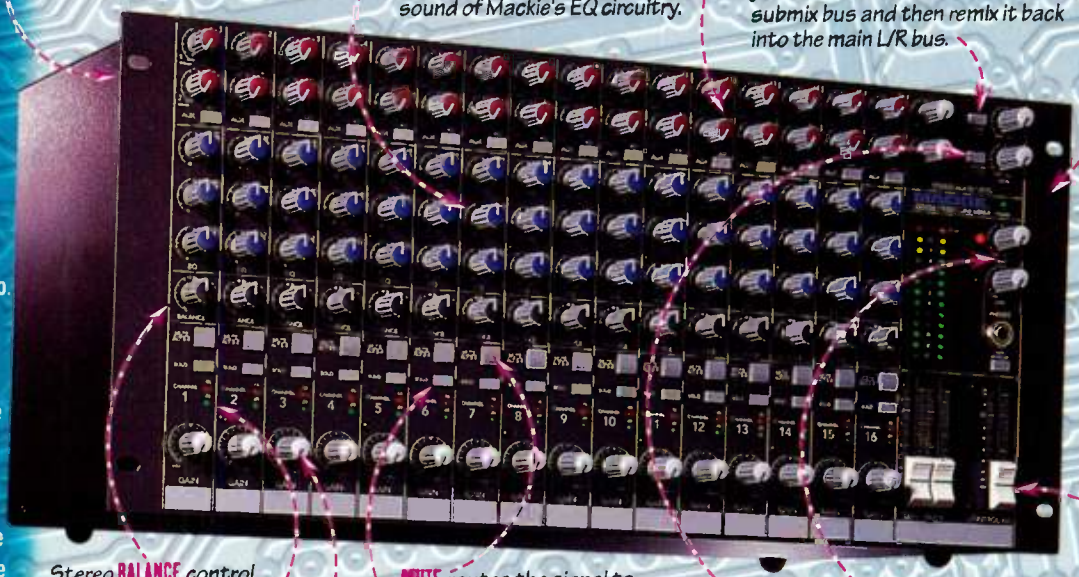
Whether you're looking for a pro-grade instrument or effects submixer, an economical main mixer for a MIDI project studio, or the perfect solution for solo or duo club acts, you've gotta check out the LM-3204. It can make a big impact on your creativity without making a big crater in your equipment budget. Call toll-free today for a free 40-page full line brochure & applications guide.

Sixteen **STEREO CHANNELS** in five rack spaces. Special mix amp architecture prevents overload from multiple hot inputs.

3-BAND EQUALIZATION at 12kHz, 2.5kHz and 80Hz. Others have copied our frequency points, but none have achieved the musical sound of Mackie's EQ circuitry.

4 **AUX SENDS** per channel (2 accessable at any one time).

ALT 3-4 TO AUX RETURN 3 switch lets you use Alt 3-4 as a stereo submix bus and then remix it back into the main L/R bus.



Stereo **BALANCE** control.

Ultra-sensitive –20dB **SIGNAL PRESENT** LEDs give you a constant visual indication of what's on every ch. Overload LEDs, too.

CHANNEL GAIN CONTROL with an extra 15dB of gain PAST Unity Gain.

INSIDE: All those goodies that set Mackie apart from the clones... double-sided thru-hole-plated fiberglass main circuit board, gold-plated internal interconnects, exceptionally high RFI rejection input design and much more.

Four **AUX SENDS** (2 stereo & 2 mono) with ultra-high gain. If you're not using effects, the 4 stereo **AUX RETURNS** can also be used as extra stereo inputs.

FINEPRINT: [1] Suggested retail. Higher in Canada. [2] Denotes useage or ownership only, as reported to Mackie Designs, and is in no way intended to represent official endorsement by the individuals or groups mentioned in this ad. [3] When eaten as part of a balanced breakfast.

MUTE routes the signal to the **ALT 3-4** stereo bus.

Stereo **IN-PLACE SOLO** maintains stereo perspective including effects; also meters individual channel level on 13-LED ladder.

SOLO & **HEADPHONE** level controls.

AUX RETURN TO CONTROL ROOM button sends Aux 4 to headphone & monitor buses so you can "wet monitor" or play along with a cue or click feed.

45mm **MASTER L/R** & **CONTROL ROOM** faders.

Built-in **POWER SUPPLY.** No wall wart!

CONTROL ROOM outputs to power amp (frees up your headphone output).

1/4" and RCA **TAPE** inputs/outputs.

Stereo **INSERTS** on Chs. 1-4.

Two of Mackie's renowned **MIC PREAMPS** with phantom power and trim controls are patchable to any input channel. Great for live sampling, acoustic or vocal tracking, small single or duo lounge acts or post-industrial speed thrash karaoke.

INPUT for **LM-3204E EXPANDER.** Basically an LM-3204 without a master section, the LM-3204E adds sixteen stereo channels and four Aux Returns in five rack spaces for $899[1].

LM-3204s are currently on tour with BoyzIIMen, Chicago, Moody Blues, King Crimson, Little Feat & other notable line-input-deprived touring acts[2].

MACKIE™

16220 Wood-Red Road NE • Woodinville • WA • 98072 • 800/898-3211 • 206/487-4337
mackie.com • Outside the US 206/487-4333 • Represented in Canada by S.F. Marketing 800/363-8855

Analog Service, Part 1

Analog synth basics and common power-supply ailments.

By Alan Gary Campbell

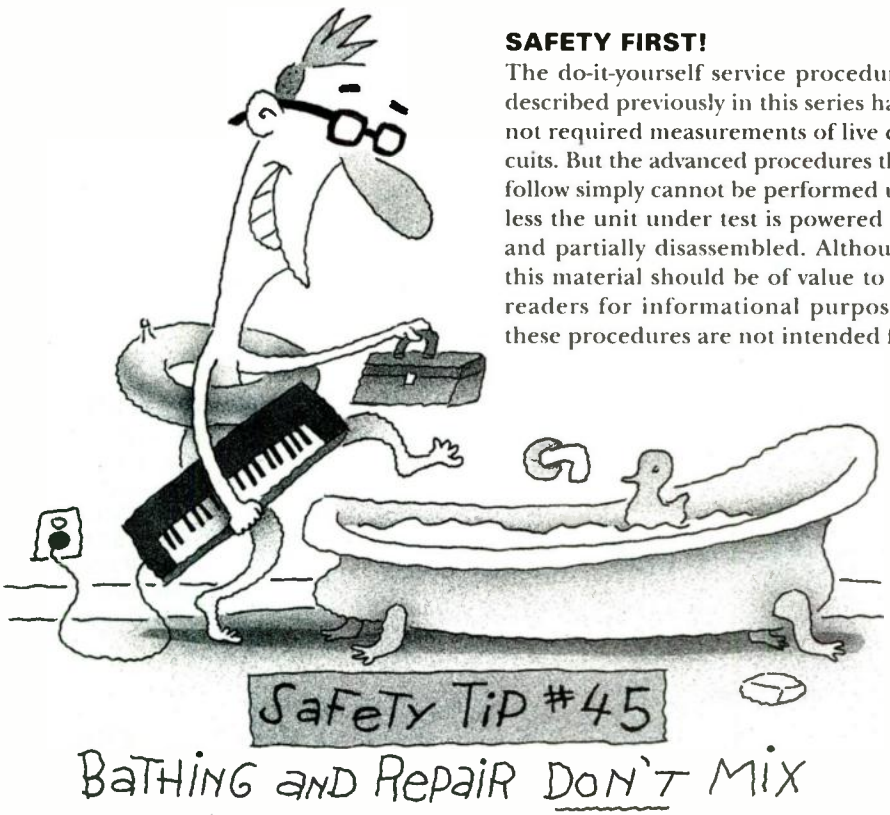
Since we began our minicourse in electronic musical-instrument service in the February 1995 issue, we have explored tools, basic DIY service, soldering and desoldering, and some "tricks of the trade." This month, we get down to the nitty gritty with the first of several columns that address analog, hybrid, and digital-synth repairs.

SAFETY FIRST!

The do-it-yourself service procedures described previously in this series have not required measurements of live circuits. But the advanced procedures that follow simply cannot be performed unless the unit under test is powered up and partially disassembled. Although this material should be of value to all readers for informational purposes, these procedures are not intended for the beginner. *AC line voltages are present at various points inside most gear and pose a serious shock hazard.*

Many manufacturers go to great lengths to insulate AC line-voltage contacts, but some don't. *Do not attempt to perform these procedures unless you have considerable knowledge and experience regarding AC line-voltage safety and proper circuit-measurement techniques.* If you are not sure how to proceed, refer the work to a qualified technician.

Note that the do-it-yourself servicer should obtain the service manual for the instrument in question and study it thoroughly before attempting circuit measurements or repairs.



JACK DESROCHER

ANALOG DEFINED

Analog synths comprise diodes, transistors, op amps, comparators, and other components that operate on continuously varying signals, which are analogous to sound waves. An analog oscillator directly produces a specific waveform; for example, a crude sawtooth oscillator can be constructed from a basic, comparator-triggered, capacitor-charging circuit. This oscillator's behavior is analogous to that of a vibrating string. An analog lowpass filter, which might be as simple as a capacitor and a resistor, can be used to control the signal's harmonic content.

In contrast, a digital oscillator is a microprocessor- or software-based construct that generates a pattern of numbers

(zeros and ones), which describe a waveform. A digital-to-analog converter transforms this computer data into minute slices of a finished waveform, which are smoothed, using filtering and other techniques, to form a listenable facsimile of the intended waveform. Timbral control for a digital oscillator usually consists of supplying different data so that the "slices" construct a different waveshape. In sum, analog technology uses continuously varying signals; digital synths use data to construct sound in discrete steps.

Understanding analog service techniques is obviously important for those who want to keep vintage gear running, but even digital equipment has significant analog components, especially in power supplies and output stages. Moreover, the building blocks of even the most complex digital ICs descend from basic, analog circuits.

BUILDING BLOCKS

The basic subcircuits of an analog synth are shown in **Fig. 1**, simplified for the sake of clarity. (Most real synths would have additional VCOs, for example, and might have such esoteric components as an envelope follower, lag processor, or sample-and-hold.) The power supply, though not shown, is the one subcircuit that connects to and affects all the others.

The aspiring tech should commit this basic signal flow and this control scheme to memory. With complex or intermittent problems, it is invaluable to be able to picture the signal or control flow in its simplest form and work backward, eliminating possibilities until simple deduction reveals the problem.

POWER-SUPPLY SERVICE

Assuming that the unit has been thoroughly checked for signs of damage or catastrophic failure (see the September 1995 "Service Clinic"), the first submodule to check is the power supply. If the power supply doesn't work right, the instrument won't work right, and damage may result. If the power supply is out of calibration, subsequent calibration of oscillators or filters may be ineffective.

Most analog synths utilize bipolar power supplies, which provide symmetric positive and negative voltages with reference to ground. This simplifies circuit design, especially in circuits that use operational amplifier ICs (op amps), and it facilitates the amplification and processing of AC (bipolar) signals. Common bipolar supply values are ±9, 12, and 15 VDC.

Some analog synths use discrete, transistor power-supply components, but

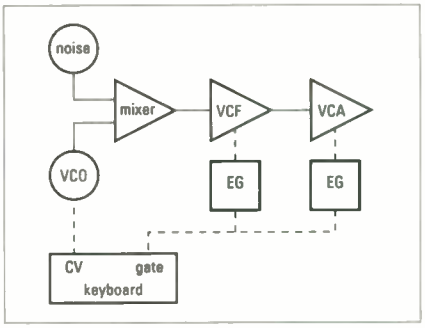


FIG. 1: In this simplified analog-synth block diagram, control sources are represented by rectangles, sound sources by circles, and output modifiers by triangles. Control voltages and triggers are indicated by dashed lines and audio signals by solid lines.

the LM7800/7900-series of 3-terminal IC regulators is more common. These devices look just like TO220-type transistors but in fact comprise a precision voltage reference, current amplifier, and output-overcurrent/thermal-overload protection circuit on a single chip. A pair (one 7800-type positive and one 7900-type negative regulator) plus a handful of support components form a complete, regulated, bipolar supply.

Regardless of the power-supply type, the output voltages should be checked under normal operating conditions before any other service is attempted. A good service manual should provide complete specifications for allowable power-supply parameters. For "ballpark" measurements, a 3½-digit, 0.1% accuracy DVM is sufficient. Generally, a deviation of five percent or more indicates a possible failure mode. When checking LM7800/7900-series devices, remember that the ground- and output-pin assignments on the negative regulator are reversed.

The most common power-supply failure modes are open (or shorted) rectifier diodes and leaky filter capacitors. Both components are susceptible to damage from line transients. Three-terminal regulators fail less frequently. Nonetheless, it is good practice to completely rebuild the affected side (both positive and negative, if needed) of a damaged supply, even if surviving components appear unscathed.

Failed diodes often show physical damage. When damage isn't obvious, diodes can be checked for forward and reverse resistance using a DMM diode-test function with the unit powered down. Next, check for voltage drops with the unit powered up.

Leaky filter caps dump excess ripple on the supply lines and cause audio hum and CV instability. Ripple measurement via an oscilloscope is best, but a DVM can measure the AC voltage "riding" on the power-supply output. Amounts should be less than 1 mV.

Low output voltage may indicate a bad regulator but could result from a shorted component elsewhere that is dragging down the supply line.

Though many bipolar supplies depend upon the untrimmed accuracy of IC regulators, which is not bad, some provide trims to set the supply voltages exactly. Calibrating a trimmable power supply requires a high-accuracy DVM with 4½-digit, 0.05% accuracy or better (e.g., Fluke 8050A or its equivalent). Check the service-manual specs. Attempting to calibrate a power supply that requires a high-accuracy meter with a less-accurate type can make matters worse.

When you recalibrate a trimmable supply, recalibrate the entire instrument. Determining whether to recalibrate a bipolar supply that is close to spec is more art than science. To cite a real-world example, when a Minimoog power supply is fairly close, I leave it alone.

Power-supply transformers are also susceptible to line transients. Common failure modes include open windings (no output) and partially shorted windings (reduced output). Transformer output specs are given in most service manuals; tolerances are usually wide. Remember: set your meter for AC volts!

EM *Contributing Editor* **Alan Gary Campbell** *is owner of Musitech.*

Reviews

E-mu ESI-32 Digital Sampler

By Geary Yelton

The cost of high-end sampling hits the low end.

A million lifetimes ago (circa 1989), when the street price of the E-mu Emulator III dipped just below ten grand, I persuaded the owner of the studio at which I was working to buy one. That EIII was a workhorse for Foley, voice-over, and sequenced recording sessions. It had 16-bit sound, 44.1 kHz sampling, and 16-voice polyphony. It also had a 5-octave keyboard, a sequencer that generated and locked to SMPTE, an 80 MB internal hard drive, and a SCSI interface. (Can you say "workstation"?) All these high-tech bells and whistles drove up the cost, of course. Its 4 MB of RAM was expandable to an enormous 8 MB for around an extra $1,000.

The E-mu ESI-32 packs a lot of sampler into a 2U rack-mount box, including 32-voice polyphony, up to 32 MB of RAM, resonant filters, and an intuitive user interface.

Today, the street price of E-mu's ESI-32 sampler isn't much more than that prehistoric EIII RAM upgrade. The ESI-32 offers 32-voice polyphony with mono samples or sixteen voices in stereo and boasts specs equal to or better than the Emulator IIIx. It samples at 22.05 or 44.1 kHz, with 16-bit resolution; includes resonant filters; and sports an intuitive user interface. All this is crammed into a 2U rack-mount box and is available at an attention-grabbing list price of $1,495. At that price, the first question that came to my mind was, "What's wrong with it?"

Well, I'm happy to report that nothing is wrong with it. However, you'll almost certainly want to add options from the start, which will increase the cost. The two biggest necessities are extra memory and the optional SCSI interface ($250). The standard 2 MB of RAM just isn't enough for serious work, so you'll need to take advantage of the unit's ability to use up to 32 MB of RAM.

Storing samples on floppies is a time-consuming hassle, so you'll also want to get the SCSI option and/or the optional, internal SyQuest 270 MB removable drive (discussed later). With or without SCSI and more RAM, though, the ESI-32 is an excellent, expandable sound module.

FIRST GLANCE

The unit I reviewed includes the SCSI option, 8 MB of RAM, and an internal 270 MB removable SyQuest drive. As with most samplers, the operating system (I had version 2.0) is stored in ROM, rather than on disk. I started out with an earlier ROM version, which E-mu updated during the review period. I found a number of commands and displays were different from the previous version, and the manual doesn't yet reflect these changes.

The ESI-32's user interface follows the tradition of previous E-mu instruments. Samples are arranged into Presets and stored in Banks. You load one Bank at a time. Each Bank can hold up to 999 samples and up to 256 Presets. A Multimode button enables MIDI reception on sixteen channels, each with its own Preset, Volume, and Pan assignments. The bright, 2 × 40 character LCD is large enough to prevent eyestrain.

Korg changes the rules:

A 64-voice synthesizer can combine the best sounds and the best price.

Forget everything you know about 64-voice synthesizers. The new Korg X5D has arrived. It gives you the 64-voice polyphony you need for the most uninhibited sequencing and layering, plus two other essential features: A surprisingly low price and a dangerous arsenal of the most sought-after sounds in the music world. The X5D features classic Korg sounds from the X-Series, M1, 01/W, T-Series and Wavestation libraries, and 100 new programs and combinations. So why be bound by the old rules? Get in front of an X5D today.

Korg X5D

Full 64-voice polyphony, for sequencing and layering to your heart's content.

Over 500 sounds from the X-Series, M1, 01/W, T-Series and Wavestation libraries.

47 effects, all fully controllable in real time.

8 megabytes of PCM memory.

Built-in interface that lets you connect directly to a computer.

KORG® X5D MUSIC SYNTHESIZER

The various functions are grouped into six Modules, which are further divided into Submodules. You use dedicated front-panel buttons to select the Modules, after which you use the numeric keypad or rotary data-entry knob to select Submodules. The cursor keys then move among parameters, which are edited with the data-entry knob or numeric keypad. The Enter button confirms your edits, and the Escape button sends you back a level.

The egg-shaped buttons are a cool piece of industrial design. The clean front panel seems to emphasize simplicity. None of the front-panel buttons serve double-duty except for the Trigger keys (described shortly). As much as I like the user interface, I am disappointed it doesn't include any macros or short cuts for jumping from one command to another without stepping through Modules and Submodules.

You can trigger sounds directly from the front panel without a MIDI keyboard. An Audition button plays a sample from the current Bank at a pitch the manual says you can define, but the manual is mistaken: it plays the sample assigned to middle C. Another button enables Trigger mode, which lets you play ten different notes from the numeric keypad. You can define the note and Velocity played by each button as well as whether the sample plays once or repeats until you press the button again. This Latch function lets you create music by looping grooves, something that should be of interest to producers of dance tracks.

Like many popular samplers, the ESI-32 has no onboard effects such as reverb or delay. The only available effect is chorusing, which can simply be turned on and off. In addition to the main stereo outputs, there's a handy pair of Sub Out/Mix In jacks that can serve as either effects send/return jacks or a pair of assignable outputs. If you need more than four outputs, though, you're out of luck. All four outputs are unbalanced, ¼-inch jacks. The ESI-32 lacks the EIIIx's arpeggiator function, and there's no graphic editing of samples or envelopes.

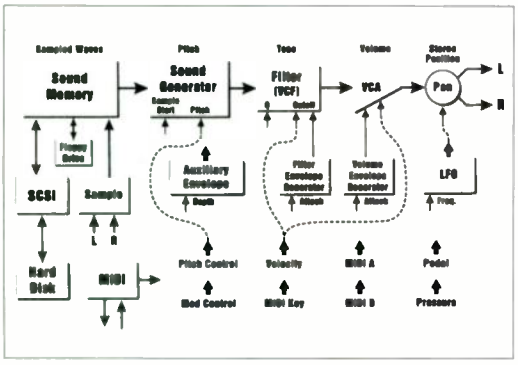


FIG. 1: The architecture of the ESI-32 is similar to that of many synthesizers.

LOADING SOUNDS

Although the ESI-32 is an excellent sampler in the Emulator tradition, there are ways to acquire sounds without sampling them yourself. The ESI-32 can import sounds via SCSI in EIII, EIIIx, Emax II, and Akai S1000 formats. Fortunately, a huge library of high-quality sounds for the Emulator III is available.

The Akai S1000 format is the most popular for importing sounds. When the ESI-32 imports Akai samples, Programs, and Volumes, any parameters that don't exist on the ESI-32 are ignored. However, the ESI-32 can do some interesting tricks to get around these limitations. For example, Akai samplers can layer multiple samples on each key, while E-mu samplers can normally play only one primary and one secondary sample per key. If more than two imported samples are assigned to a single key, a Full Placement function creates up to four Presets and links them all to be played by that key, ignoring Velocity switching and crossfading. If you turn Full Placement off, only the first two samples assigned to each key are imported. Another useful function, Adjust Loops, corrects any sample-looping problems that crop up during translation.

You can also import sounds into the ESI-32 via SCSI Musical Data Interchange (SMDI). Although SMDI doesn't import Banks, Presets, and most parameter information, it does import raw samples and keymaps along with pitch information. Using Passport's *Alchemy* 3.0, I was able to transfer samples quickly to and from the sampler with ease. I never encountered any problems in this process.

SONIC MANIPULATION

The architecture of the ESI-32 resembles that of many synths (see **Fig. 1**). Each voice includes a resonant, lowpass filter; highpass and other filter modes aren't available. You also get three attack-hold-decay-sustain-release (AHDSR) envelopes and one multiwaveform LFO per voice. You can assign up to six MIDI messages to control any of nine internal parameters in real time (see **Fig. 2**).

Pitch and VCA level and attack time can be modulated using Velocity, and several parameters can be modulated in real time with MIDI controllers. There are no footswitch or pedal jacks on the ESI-32. Portamento lets you glide from one note to another at a user-defined rate.

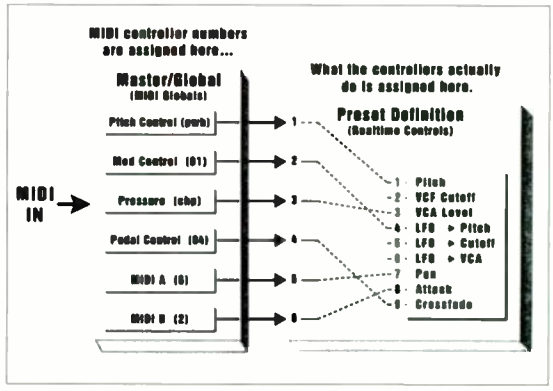


FIG. 2: You can assign up to six MIDI messages to control nine different parameters in real time within the ESI-32.

The ESI-32 can automatically normalize and truncate samples when they're recorded, if you like. You can also predefine the keyboard range; as soon as one sample is recorded, you're ready to record another. All samplers should have such time-saving features.

The ESI-32 offers flexible looping capabilities, although only one forward loop is available per sample. However, the loop can play during the release phase of the VCA envelope. You can also monitor the sound of the loop as you change its start point and size, which is convenient. The Auto Correlation function finds the best loop-start and -end points closest to the points you specify. Repeatedly pressing the left or right cursor key moves from one potential loop point to another, but holding the button doesn't scroll.

Loop Compression compresses everything between the loop points, minimizing changes in amplitude over the

course of a loop. You can also crossfade loop points. In addition to equal-power crossfade looping that suppresses dramatic shifts in level, linear crossfade looping fixes loops that are not quite perfect.

As expected, you can cut, copy, and paste portions of sounds either within a sample or from one sample to another. The selected portions are temporarily backed up to disk as these editing operations are performed. If you have enough hard-disk space for such backups and the disk isn't locked, you can use the Undo and Redo commands to cancel or restore your last cut, paste, crossfade loop, sample-rate conversion, or other editing operation. I wish there was a physical Undo button on the front panel, though; it's too cumbersome to go to the Digital Processing Submodule and select the Undo Submodule every time you want to undo something.

Some of the ESI-32's most interesting processing functions are found in the Digital Tools II Submodule. For example, sample-rate conversion is continuously variable from 7 to 50 kHz. The 1-band parametric equalizer boosts the specified frequency band in the selected portion of the sample by up to +12 dB or cuts it by as much as -48 dB. It is a fully parametric EQ, meaning you can define the center frequency and bandwidth.

Using time compression, you can also change the length of a sample without changing its pitch. There are ten types of compression or expansion that depend on the type of sample being processed. Sample types range from deep

Product Summary

PRODUCT:
ESI-32 digital sampler
PRICE:
$1,495
MANUFACTURER:
E-mu Systems, Inc.
1600 Green Hills Rd.
Scotts Valley, CA 95067
tel. (408) 438-1921
fax (408) 438-8612
CIRCLE #437 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ●
EASE OF USE	● ● ● ◖
SOUND QUALITY	● ● ● ● ◖
VALUE	● ● ● ● ◖

(bass) to tight (drum loops) to noisy (nonpitched). If one type of time compression doesn't sound right and you have the patience, just undo it and try another. On the other hand, Pitch Change alters a sample's pitch without changing its length. Again, the same list of ten sound types is available. Pitch can be shifted by as little as one cent or as much as one octave up or down, and the results sound quite good.

Transform Multiplication merges two samples in a unique way. The frequencies that both sounds have in common are retained while other frequencies are discarded. The final product of this function is a sound with a reinforced harmonic spectrum. Like time compression, it takes a lot of time, and once started, there's no graceful way to interrupt the process.

As you would expect, sounds with lots of common harmonics yield sounds with the most harmonics. If their harmonics clash, the result sounds quite thin. Using Transform Multiplication, I was able to create some really ugly new timbres in only a few hours. Until I've developed the ability to predict the results better, creating prettier sounds may take a few days.

I had hoped that Doppler/Pan, a 2-dimensional localization function, would be one the most interesting features of the ESI-32. Using a combination of pitch shifting and stereo panning, Doppler/Pan creates the illusion of moving not just left and right but also forward and backward in the 2-dimensional listening space. You can choose from several preset paths or define your own path along which the sound "moves." The Path Management screen lets you define the *x* and *y* coordinates for up to 26 different points in the path and the time it takes the sound to move from one point to the next. The coordinates are expressed in tenths of a foot, with a maximum distance of 99.9 feet in any direction on a flat plane.

This sounds great in theory, but the results were not convincing, except when the path was simple and the sample was compressed or of relatively constant volume. It's easy to get lost when experimenting with Doppler/Pan, and defining a user path is tedious.

OPTIONS, ANYONE?

The baseline ESI-32 sampler comes with two 1 MB SIMMs mounted in 30-pin SIMM slots. To increase the memory, you must pull out these SIMMs and install another matched pair. Using 4 or 16 MB SIMMs, you wind up with 8 or 32 MB of RAM. Four SIMM slots and a maximum of 64 MB would be better, but I'm not complaining.

You should probably get the SCSI upgrade when you buy the unit. This provides a 50-pin port on the back panel. As there's only one port, the sampler must be at one end of a SCSI chain.

Unlike some samplers, the SCSI ID number is easily changed in the SCSI Setup screen. The ESI-32's SCSI implementation allows up to seven "master" devices to control the same SCSI bus, which lets multiple samplers share the same CD-ROM or hard drive with a minimum of conflicts. A Disk Utilities parameter lets you tell the ESI-32 to avoid a host with any SCSI ID you specify. This is sometimes necessary for coexisting with Macintosh computers in the same SCSI chain.

If you decide to get the model with

an internal 270 MB SyQuest drive, you sacrifice the floppy-drive port, but you gain a large starter library of excellent sounds that half fills a removable cartridge. Of course, without a floppy drive, you can't load sounds directly from a floppy disk. I encountered no problems with the SyQuest drive, except that it occasionally stopped for a few moments when loading certain Banks while the ESI-32 was connected to the Mac via SCSI. It looked like both the sampler and the Mac were frozen. After a few seconds, though, it completed the load normally.

The ESI-32 with a SyQuest drive costs $1,000 more than the same model without it. That's twice the going price for a 270 MB SyQuest drive from other manufacturers. I'd rather buy an external SyQuest and keep the floppy drive.

Another option is the S/PDIF digital-audio interface ($200). This lets you exchange audio data with DAT machines, hard-disk recorders, and other devices with S/PDIF ports. With the digital I/O board installed, both analog and digital outputs can be used at the same time. AES/EBU ports aren't available.

CONCLUSIONS

The price/performance ratio of the ESI-32 is remarkable. The sound is excellent, and the ease of use is outstanding. The SCSI option worked flawlessly, and the removable drive is a much more flexible option than a fixed hard drive. If you've always admired Emulators you couldn't afford, here's your chance to join the pack.

The paucity of assignable outputs may be a serious problem for some users. If you need more than 32 MB of RAM, this device won't fill your needs, but then again, not many instruments will. And if you want to do a lot of fancy processing as though your sampler were a synthesizer with massive matrix modulation, you should probably look at the Emulator IV or e-64. There's no keyboard, no sequencer, no direct-to-disk multitrack recording, and no effects to speak of. It's not a workstation; it's a sampler. If you need a sampler and this one's in your price range, the ESI-32 should be a major contender for your affections.

Even after a decade of writing for **EM**, *it still takes* **Geary Yelton** *just as long to read a manufacturer's instruction manual as it takes you.*

Symetrix 488 and A.R.T. MDM-8L

By Michael Cooper

Are octal dynamics processors a "must-have" for digital recording?

With tens of thousands of Alesis ADATs, Fostex RD-8s, and TASCAM DA-88s sold worldwide, the modular digital multitrack (MDM) is here to stay. For those who have bought into the digital revolution, the familiar problems of analog tape are a thing of the past. But digital has its problems, too. If your recording level is too low, your tracks will suffer quantization noise from low bit resolution. Record too hot, and your digital recording can quickly take on the sonic characteristics of a kitchen garbage disposal.

Two new products attempt to solve these problems. The Symetrix 488 DYNA-Squeeze and A.R.T. MDM-8L are octal (8-channel) dynamics processors that compress your tracks as they get recorded to tape. The purpose behind these two boxes is to mildly tame peaks in your program material that would otherwise exceed the 0 VU "crash-and-burn" mark on your MDM, allowing you to record at hotter levels without clipping. Low-level passages are raised so that more digital bits will be used, resulting in a fuller, cleaner sound with less background hash.

The DYNA-Squeeze and MDM-8L approach these tasks quite differently, so a real-world comparison of the units is of particular interest. First, let's examine each unit; then we'll let them duke it out in some studio sessions.

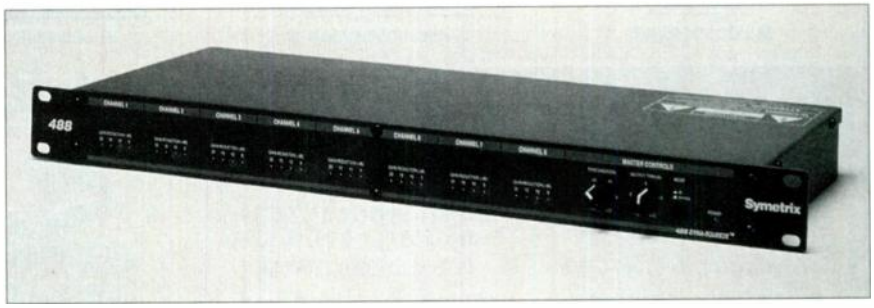
SYMETRIX 488

The DYNA-Squeeze features eight somewhat-independent channels of compression housed in a 1U rack-mount metal chassis. I say "somewhat independent" because although the channels are not linked, they share the same three master controls: a threshold knob, an output trim knob, and a bypass switch. (In a linked stereo compressor, both channels respond to the dynamics changes on the master channel, which is quite different from simply sharing a single set of parameters, as in the DYNA-Squeeze.) This dearth of controls constrains all eight channels to the same settings but also makes the DYNA-Squeeze a very straightforward, easy-to-use device.

The reason behind this spartan setup is that Symetrix never intended the DYNA-Squeeze as a channel-insert compressor for use on individual tracks. Rather, it is meant to maximize your MDM's headroom by serving as an in-line device between the mixer's multitrack group outputs and the recorder's inputs. Although Symetrix also espouses the DYNA-Squeeze's use for analog recording and live sound (to prevent overload of stage-monitor systems, for example), the owner's manual focuses almost exclusively on digital-recording applications.

Compressor attack and release times are preset for all eight channels at approximately 1 millisecond and 1 second, respectively. The DYNA-Squeeze processor utilizes soft-knee compression at a preset 2.5:1 ratio. Each compressor channel features its own 4-segment LED bar-graph meter, showing the amount of gain reduction in 5 dB increments, up to 20 dB.

The wide-ranging master threshold control sets the threshold level for all eight channels between -40 and +10 dB.



Symetrix's 488 DYNA-Squeeze provides eight channels of compression for maximizing an MDM's headroom. The unit's sound is reasonably transparent, but it has only a few global controls, and the channels can't be bypassed.

freedom of choice

Up to 31 Voices

Up to 16 MB Memory

Stereo Sampling,
Sampling through Effects,
Resampling from Sequencer

24-Bit Effects

16-Track Sequencer

Digital Audio Recording

SCSI

Optional Digital I/O

make music think ENSONIQ

your newest choice: the ASR-88

When it comes to samplers only ENSONIQ offers you a choice of models to fit your performance and recording needs.

The 61-key synth action ASR-10 combines advanced sampling, sequencing, and our Poly-Key™ pressure keyboard. The ASR-10 rack is ideal for studio or live performance. And now – the 88-key weighted action ASR-88. The feel and range of a true piano keyboard married to all the features that only an ASR delivers.

Whichever you choose, you'll get the most powerful sampling and music creation tool ever. With a huge library of sounds on floppy disk and CD-ROM, including our new Signature Series CD's from Chicago, D.J. Jazzy Jeff, Steve Gadd, and others. Plus a world of sounds, ready to play, from Akai™ and Roland™ CD's.

The ASR Series are the only samplers to combine sampling, effects, sequencing, and digital audio recording in one musical instrument.

Whatever your style of music, however you play it, you now have the freedom to choose. So what'll it be?

ASR-88 Bundle
16 MB Memory
SCSI Installed
CD-ROM Drive
CDR-3 & CDR-4 Libraries

ENSONIQ
LEADING THE WORLD IN SOUND INNOVATION

☐ Tell me more about my ASR choices!

Also send info on:
- ☐ KT Series *64-voice weighted action synths*
- ☐ TS Series *synths*
- ☐ KS-32 *weighted action synth*
- ☐ SQ Series *synths*
- ☐ DP/4+ *effects processor*

☐ I'd also like your *make music* **think ENSONIQ** demo CD *(here's $5; ship it, and watch how you handle it!).*

Name _____

Address _____

City _____ *State* _____ *Zip* _____

Phone (_____) _____

Mail to: ENSONIQ, Dept X-56 155 Great Valley Parkway
P.O. Box 3035, Malvern, PA 19355-0735

Every ENSONIQ instrument is made in America. To reach us: 800-553-5151 phone, 610-647-8908 fax 800-257-1439 documents to your fax GO MIENSONIQ *on CompuServe* http://www.ensoniq.com *on the Web*

Test Drive A Real Performance Machine!

Come on you control freaks. Don't just kick the tires and walk around it.

Do It!

Grab the controls.

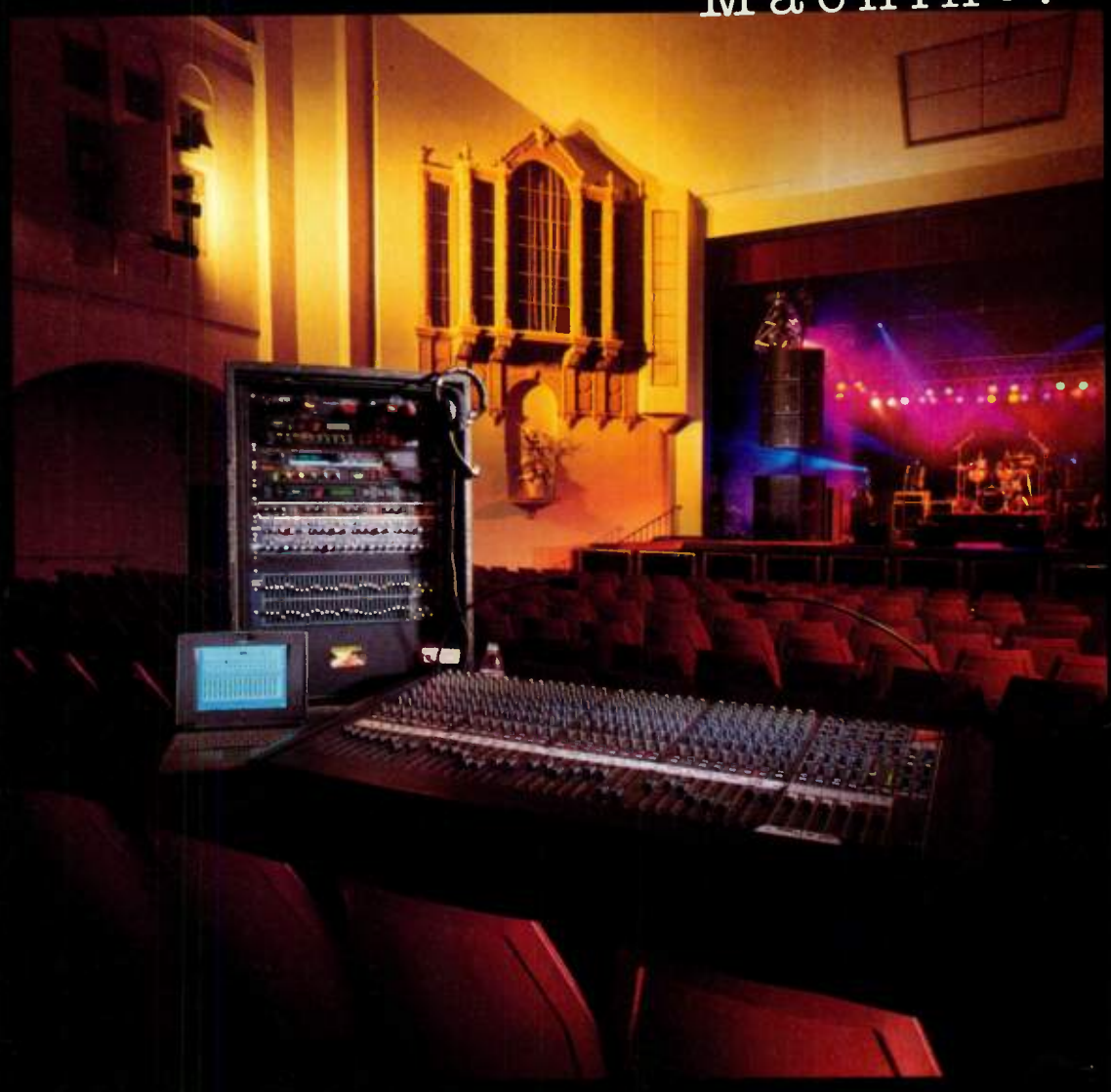
Drive it hard, to the point you fear it's going to come out of the groove like the others.

But it doesn't.

It hangs right with you begging for more.

Throw it around the curves, easily gliding out of the rough and right into the sweet zone.

And when you stop - it's silence. Pure and beautiful.



You look around the controls. Everything is laid out neat and logical - right where you want to put your hands. Nothing superfluous, just the basics needed for professional driving. Every control is silky smooth and even the switches light. And while it's easy to drive, this elegant application of high tech allows you to explore places you've never been before. It's the perfect companion for a home project room - but it's equally at ease taking abuse on the road in front of a crowd. It's expandable too, if you need to carry more inputs or more buses. You can even order up a complete option package, allowing you to customize it for your tastes.

Then the light bulb above your head dawns and the adrenaline kicks in. This is NOT like the other machines.

This one lets you connect to your music like never before. Push it, feel it, move it, drive it - easily and artfully. Great sound, control, and driving excitement. After all, isn't that what this is supposed to be about? And when the salesperson tells you it starts at a base sticker price of under $2000*...

But then again, what else would you expect from the company that helped invent rock & roll?

The MX-5200 mixing console system from Fender Pro Audio (available in 16, 24, and 32 input versions, plus options).

Fender® ELECTRONICS

*M.S.R.P. for 16 channel version. Does not include applicable local taxes, licenses, or dealer prep charges. Concert rig courtesy of Spectrum Audio. ©1995, F.M.I.C. For more information, send $2.00 postage and handling to: MX-5200, c/o Fender Pro Audio, 7975 N. Hayden Rd., Scottsdale, AZ 85258.

WRH

The master output-trim knob applies -10 to +10 dB of make-up gain (i.e., post-compressor gain) equally to all eight channels. More gain would have been helpful in certain situations. For instance, say you have an analog master tape that was recorded at wimpy levels, and you want to transfer it to MDM via the DYNA-Squeeze. If the 488's output trim were heftier, you could crank it enough to get the tracks up to a healthy level and avoid having to run the tracks through an additional intermediary stage of amplification (e.g., the mixer's preamps). For most applications, the 488's output gain is sufficient, though it clearly does not take full advantage of the unit's respectable +20 dBu maximum output-level specification.

The DYNA-Squeeze uses only two op amps and one VCA per channel, which contributes to a clean signal path. Frequency response is stated as 20 Hz to 20 kHz, +0/-1 dB; this is doubtless an "unprocessed" spec. Signal-to-noise is rated at better than 111 dB, 20 Hz to 20 kHz, and indeed, I found the unit to be very quiet. Even the channel closest to the power-supply transformer (channel 8) exhibited virtually no hum.

Rounding out the DYNA-Squeeze's front panel are a mode switch that enables/disables all eight compressors and an LED that indicates the unit is powered up. The mode switch is not a hardwire bypass and lacks a status LED. There is no power switch for the unit; plug it in, and it's on.

Product Summary

PRODUCT:
488 DYNA-Squeeze
8-channel
compressor/interface
PRICE:
$579
MANUFACTURER:
Symetrix
14926 35th Ave.
W. Lynnwood, WA 98037
tel. (800) 288-8855
or (206) 787-3222
fax (206) 787-3211

CIRCLE #438 ON READER SERVICE CARD

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

All connections are on the rear panel. Separate TRS phone jacks are provided for each channel's input and output. Inputs are balanced, though they accept unbalanced signals with no problems. The output connectors are unbalanced, with the ring connected to circuit ground, but they also work with balanced signals.

The DYNA-Squeeze's rear panel -10 dBV/+4 dBu output-level switch allows you to interface +4 dB consoles with -10 dB tape machines. Pushing the switch in (-10 dBV) drops the output 11.8 dB, permitting you to patch into an ADAT's phone jacks without overloading the A/D converters. If your console's nominal output level is -10 dBV, you should leave the switch out (+4 dBu). Rounding out the 488's rear panel is a detachable, 3-prong AC cord. Thankfully, the power supply is internal.

A.R.T. MDM-8L

The MDM-8L offers more features and controls than the DYNA-Squeeze, making it at once more versatile and harder to set up. Like the DYNA-Squeeze, the 1U rack-mount MDM-8L features eight channels of soft-knee compressors that are preset at a 2.5:1 ratio. But A.R.T.'s box also features an additional mode of operation (activated via the front panel MDM Mode switch) that switches in a separate 20:1 peak limiter after each compressor. This all but guarantees you will never exceed the 0 dB digital record level. The limiter's threshold is preset at 6 to 8 dB higher than the compressor's threshold.

Each channel also boasts separate input- and output-level knobs, and each has its own hardwire-bypass switch (with associated LED), allowing far more independent operation than is afforded by the DYNA-Squeeze. Furthermore, channels 2 through 8 each feature a Link switch (with associated LED) that, when depressed, links each respective channel to the first unlinked channel to its left. In this manner, multiple contiguous blocks of channels can be linked. By linking two channels, I compressed an acoustic guitar (miked in stereo) without the image shifting. Depressing all seven Link switches causes all channels to track channel 1's gain reduction.

On the down side, the unit lacks the helpful gain-reduction meters included with the DYNA-Squeeze, although a separate threshold LED is provided for

ADVERTISER INDEX

Advertiser	Reader Service #	Page
Ace Music Center	-	121
Ace Products	501	141
Acoustic Sciences Corporation (ASC)	502	101
Alesis (QS6)	503	16-17
Alesis (Q2.0)	504	123
Allen & Heath	505	33
Aphex Systems	506	23
Applied Research & Technology (A.R.T.)	507	97
Artic Software	508	94
Ashly	509	43
Audio-Technica	510	73
Bananas at Large	511	145
Beatboy Drum Sequences	512	111
Big Briar	513	149
Caruso Music	514	142
CMCI	515	50
Computers & Music	516	126
Cool Shoes	517	115
DGS Pro Audio	518	158
Digidesign	-	70-71
Disc Makers	519	57
Discount Distributors	520	157
Drexel University	521	115
Ebtech	522	143
Electro-Voice (EV)	523	76
EMAGIC	524	60-61
E-mu Systems	525	75
E-mu Systems #2	526	89
Ensoniq (DP/4+)	527	7
Ensoniq (ASR-88)	528	135
Europadisk	529	51
Eye & I Productions	530	53
Fender	531	136
Fostex	532	14-15
Greytsounds	533	96
Hafler	534	26-27
Howling Dog Systems	534	145
Hughes & Kettner	536	93
Imaja	537	158
JBL Professional	-	99
KAT	538	113
Key Electronics	-	88
Key Electronics #2	-	132
Korg (Trinity)	539	12-13
Korg (Prophecy)	540	81
Korg (X5D)	541	130
Kurzweil Music Systems	542	171
L & M Music	543	153
Leigh's Computers	544	149
Lexicon	545	59
Los Angeles Recording Workshop	-	147
MacBEAT	547	143
Mackie Designs (CR-1604)	548	2-3
Mackie Designs (LM-3204)	549	124
Macromedia	550	44
Mark of the Unicorn	551	172
MIDIMAN (MM-401/Macman)	552	95
Mix Bookshelf	553	161
Music Supply	554	109
Musician's Friend	555	72
nord lead/Armadillo Enterprises	556	92
Oberlin College Cons. of Music	557	155
Opcode	558	29
Opcode #2	559	68
Opcode/Music Quest	560	52
Opcode/Music Quest #2	561	108
Peavey Electronics	562	40-41
PG Music (Band-In-A-Box)	563	78-79
PG Music (Power Tracks Pro)	564	127
Polyline	565	157
PreSonus Audio Electronics	566	69
QCA	567	151
QSC Audio Products	568	35
Quik Lok/Music Industries	-	84
Rack Accessories	569	87
Rhythm City	570	147
Rich Music	571	102
Roland (MS-1)	572	11
Roland (DM-800)	573	66-67
Sam Ash Professional	-	133
Samick	574	104
Shure	575	46
Sonic Foundry	576	118
Sony	-	20-21
Sound Quest	577	137
Soundspiration	578	159
SoundTech	579	103
Soundtrek	580	122
Soundware	581	85
Speir Music	582	132
Spirit	583	65
Steinberg/Jones	584	47
Strings & Things	585	151
Studiomaster	586	39
Sweetwater Sound	587	117
Switchcraft/Raytheon Co.	588	129
Symbolic Sound	589	100
TASCAM	590	25
Taxi	591	112
Technics	592	30-31
Tech 21	593	98
Thoroughbred Music	594	101
Tune 1000 Corporation	595	102
Twelve Tone	596	8-9
Vinylstyle	597	107
Voce	598	62
Waves	599	153
West L.A. Music	600	141
Wildcat Canyon Software	601	52
WinJammer	602	114
The Woodwind & The Brasswind	603	114
World Records	604	155
Wurlitzer	605	74
Yamaha	606	106
Yorkville	607	119
Zefiro Acoustics	608	142

RATE THE ARTICLES IN THIS ISSUE!

OCTOBER 1995

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. "Production Values: Audio Visionary," p. 36	701	702	703	704
b. "Creative Space: Wright at Home," p. 48	705	706	707	708
c. Cover Story: "Power Tools," p. 54	709	710	711	712
d. "Lip Service," p. 82	713	714	715	716
e. "Flash in the RAM," p. 90	717	718	719	720
f. "Working Musician: Networking with E-Mail," p. 110	721	722	723	724

each channel to indicate when the signal is above threshold and compression is taking place. Maximum compression is 50 dB.

Separate balanced, ¼-inch TRS, input and output jacks are provided for each channel on the MDM-8L's rear panel. These also accommodate unbalanced signals and tip-sleeve connectors.

The MDM-8L's independent-channel operation lets you connect it to your mixer's multitrack subgroup outputs for maximizing the MDM's headroom, to mixer inserts for processing tracks individually during recording or mixdown, between a sound-reinforcement console and a crossover or power amplifier, or to multiple systems simultaneously.

A detachable, 3-prong AC cord and -10 dBV/+4 dBu switch (for setting nominal output levels in MDM Mode) round out the rear panel. A front-panel LED lights when +4 dBu output is selected. Thankfully, a power switch (with LED indicator) is also included on the front panel.

As with the DYNA-Squeeze, the MDM-8L's compressor attack time is 1 ms (in MDM Mode, the limiter's attack time is 100 µs). However, the release time is program dependent, varying between 50 ms and 2 seconds. The channel-output knobs boast twice the range of the DYNA-Squeeze and can adjust make-up gain from -20 to +20 dB. The threshold is also a bit more adjustable, from -40 to +20 dBu. Maximum output level is comparable to the DYNA-Squeeze and is rated at +21 dBu.

The unprocessed frequency response is stated as 10 Hz to 30 kHz, ±1 dB. The unit's broadband output noise is rated at -95 dBu. Indeed, all channels were quiet and exhibited virtually no hum, although my review unit produced an audible mechanical vibration caused by sympathetic resonance between the internal power supply and the unit's top panel. According to A.R.T., this has been fixed in current production units. The MDM-8L carries a generous 5-year warranty (versus Symetrix's 1-year warranty for the DYNA-Squeeze).

Lest you be completely blinded by the MDM-8L's extensive feature set, I should tell you that it's not all a bed of roses. Operating the unit can be confusing because of the design and labeling of controls. For instance, in normal/compressor mode, turning the input knob clockwise does *not* raise the input level but instead lowers the threshold for that channel. In MDM (Limiter) mode, turning the input knob clockwise lowers the threshold while proportionately *raising* the output level. (The output knob is inactive in MDM/Limiter mode.)

This inverse relationship and the lack of dedicated controls for threshold level and make-up gain make it impossible to apply post-compressor gain without overprocessing the input signal. Alternatively, if you feed the unit's front end more gain from your mixer, raising the threshold level to prevent overprocessing will result in a reduced output/record level.

HEAVY-COMPRESSION TESTS

Given that the MDM-8L lacks gain-reduction meters, setting up both units for equal levels of compression involved some approximation. For the following A/B/C tests, I set the output-level controls of both units for about the same amount of make-up gain (with no compression, initially) and then lowered the thresholds so that output levels decreased by an equal amount. All vocals and instruments were recorded through a preamp, the output of which was then multed three ways.

One output went directly to track 1 of an ADAT, unprocessed. The second output was fed through the DYNA-Squeeze and recorded onto track 2. The third output went through the MDM-8L and onto track 3. Switching between the three tracks on playback allowed me to make A/B/C comparisons between the two processed signals and the "dry" recording.

My first tests involved recording vocals with heavy compression. Note that this is not how you would actually use the unit; the sole purpose of this test was to let me hear the quality of the VCAs in a torture-test setting. As with virtually any compressor, very heavy compression yields a dull sound, raises the ambient noise floor of the audio signal, and can cause pumping and breathing artifacts. The DYNA-Squeeze and MDM-8L were no exceptions.

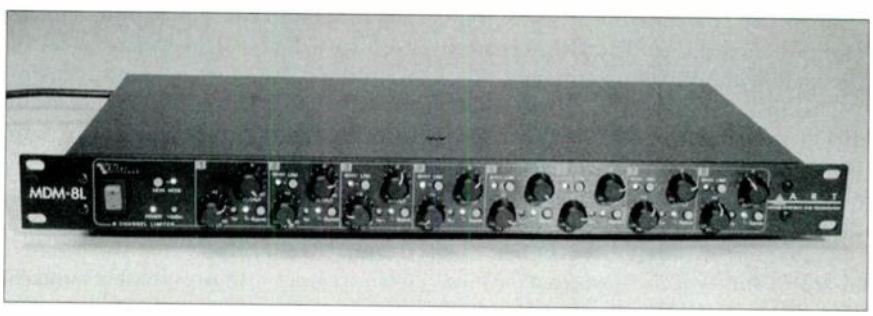
However, in my heavy-compression VCA tests, the DYNA-Squeeze proved to be much more transparent than the MDM-8L, which exhibited far more pumping and breathing, accentuated sibilance more, and lost much more warmth and body. Clearly, both units are meant to be used for only light compression, as a safety net, and the DYNA-Squeeze's gain-reduction meters make this easier to set up.

The bypass switches on both units work silently, introducing no dangerous pops or clicks. When the DYNA-Squeeze is bypassed, it restores unity gain slowly, which is a nice feature.

LIGHT-COMPRESSION TESTS

For my next series of tests, I set up both units for very mild compression so that the DYNA-Squeeze's first gain-reduction LED and the MDM-8L's threshold LED both lit only on peaks. All instruments and vocals were multed to three tracks (six tracks for instruments miked in stereo) of an ADAT, as in the previous tests. The heavy-compression tests were designed to evaluate the VCAs, but this time I set the units up for regular real-world applications.

First, I recorded a strummed acoustic guitar with a pair of B&K 4011 microphones through a Millennia Media HV-3 mic preamp. (These high-quality components delivered a pristine signal for my tests.) While both the DYNA-Squeeze and the MDM-8L added some fullness to the guitar, the compression also made the guitar a little boomy in the upper bass region. Mild pumping



A.R.T.'s MDM-8L offers eight independent compressor/limiters with more features and controls than the DYNA-Squeeze. However, it lacks gain-reduction meters and colors the sound noticeably.

(amplitude modulation) was also evident on full chords containing low bass notes. Furthermore, the transient attack on the guitar was squashed, making the track sound less lively.

Next, I recorded a full drum kit. Recording the kick drum with an AKG D112 mic through a robust API 512b mic preamp, I fine-tuned the threshold controls of both units for roughly equivalent background bleed. Both the DYNA-Squeeze and the MDM-8L robbed the API of much of its warmth and bass punch, thinning out the kick and simultaneously raising the level of the snare bleed into the mic.

The snare-drum tracks (recorded with a Shure SM57 and the API 512b) were equally disappointing. The DYNA-Squeeze rounded off the nice, sharp attack and dulled the sound overall. On a brighter note, it kept the release phase of the envelope intact. The MDM-8L, on the other hand, dramatically emphasized the snare drum's release phase so that there was an excessive increase in the rattle of the snares. Although the MDM-8L made the snare drum sound brighter, it nevertheless had less impact. Both units robbed the snare drum of some low-end punch and significantly increased the kick-drum bleed into the mic.

The hi-hat (recorded with a B&K 4011) did not fare well, either. The DYNA-Squeeze made it sound choked and dull. Pumping and an unwanted increase in room ambience were also obvious. The MDM-8L also robbed the hi-hat of some of its upper harmonics and choked the envelope a bit, but compared to the DYNA-Squeeze's recording, the inherently bright timbre was better preserved overall, and pumping was less evident.

Finally, I recorded male vocals with a Neumann U87A and the HV-3 preamp. Both units did well with light compression. No pumping or breathing was evident. The vocal was bigger and fuller sounding, with the MDM-8L adding a tad more presence than the DYNA-Squeeze.

In the interest of comparing "apples to apples" in my tests, the MDM-8L was A/B'd against the DYNA-Squeeze using the A.R.T. unit's Normal (compressor) mode. In separate tests, the MDM mode proved to be harsher sounding than Normal mode.

Just to make sure I was using both units as the manufacturers had intended, I contacted Symetrix and A.R.T. to compare notes. To my surprise, Symetrix's representative told me to ignore the setup specified in their excellent owner's manual, which generally recommends using approximately 5 to 10 dB of compression or less. He suggested using considerably higher compression so that the audio would not continually cross over the threshold and cause pumping. However, when I retested the unit as instructed, the higher compression squashed and thinned out the sound on every instrument even more and produced unacceptable background bleed and pumping.

CONCLUSIONS

Compressors with preset attack and/or release parameters are bound to work better on some instruments than on others. Both the DYNA-Squeeze and the MDM-8L performed well with light compression on nonpercussive sources, such as vocals. For this reason, I suspect that saxophone, flute, and organ would also be no problem for both of these units. However, neither unit handled the broad, complex spectral content of acoustic guitar consistently well (that is, without pumping occasionally). On percussive instruments with a lot of bottom end (kick drum), transient punch (snare drum), or complex, high overtones (hi-hat), both units failed miserably.

The MDM-8L allows you to bypass individual channels (for example, to avoid compressing drums and acoustic guitar), a major advantage over the DYNA-Squeeze, which is all or nothing, with only a global bypass. A.R.T.'s unit also offers fully twice as much make-up gain as Symetrix's, which comes in handy for bringing low-level signals up to a beefy level after compression. But although the MDM-8L is much more versatile and flexible, the DYNA-Squeeze generally sounds better with moderate compression and is far easier to set up.

Ultimately, one has to ask whether we should even use these kinds of devices. Recognizing that a chain is only as strong as its weakest link, what do we gain by indiscriminately sticking budget compressors on all channels of a quality MDM? The question is even more pressing when, as with the DYNA-Squeeze, you have very little control over the processing. Are the artifacts that these units introduce (timbral coloration, loss of detail, pumping and breathing) worth the improvements in headroom and bit resolution? Considering that only very mild compression yields acceptable results, you're only gaining a few dB of headroom at best, which translates into less than one bit of resolution gained.

It's true that clean, sustained signal sources such as vocals, flute, and organ distort horribly when they "red light" (clip). But, contrary to popular myth, *transient hits* that barely exceed a digital recorder's 0 VU mark rarely produce audible distortion, because of their instantaneous nature. I don't worry about kick and snare drums, slap bass, and distorted electric guitars *lightly* kissing the red meter. So then, on how many tracks is completely avoiding the 0 VU mark all that critical? Do you *need* a compressor for every track? I think not.

Music production is not merely a process of coaxing levels up to the nominal level. Engineers and producers constantly make decisions throughout the process to fashion the right ambience, timbre, *and envelope*. That's what "fat," "punchy," "in your face," "round," and a host of other trade phrases are all about. Compression changes the envelope of a sound. To auto-compress all tracks in order to achieve greater headroom and bit resolution is to rob yourself of the control over one of the most fundamental (and least understood) aspects of recording and to homogenize your recordings. You wouldn't run all your analog synthesizers through the same envelope-generator setting, would you?

Finally, one of digital recording's

Product Summary

PRODUCT:
MDM-8L 8-channel limiter
PRICE:
$599
MANUFACTURER:
A.R.T.
215 Tremont St.
Rochester, NY 14608
tel. (716) 436-2720
fax (716) 436-3942
CIRCLE #439 ON READER SERVICE CARD

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

strengths over analog is its lightning-fast transient response. Transients add impact and detail to a recording and, by virtue of their high-frequency content, help the listener to localize sounds in the stereo field. Squashing transients in the interest of maximizing headroom robs digital recordings of this great advantage.

Is there any situation where you would want to compress everything at all MDM inputs? Yes; in some cases, when recording live shows where levels can change suddenly and unpredictably, you may want to protect your recording with peak limiting on all tracks. But this problem should not arise in the studio. For many tracks, *any* amount of compression is musically inappropriate. For those tracks that do need compression, a comparably priced, full-featured, 2-channel compressor/limiter (available from A.R.T. and Symetrix, among others) would be a better choice than the DYNA-Squeeze or MDM-8L.

Michael Cooper *is a recording engineer, producer, and owner of Michael Cooper Recording in Eugene, Oregon.*

Dauz Drum Kit

By George Petersen

Quality percussion pads at a bargain price.

When musicians talk about electronic percussion, Dauz is rarely the first manufacturer mentioned. Nevertheless, you've probably heard your share of music played on Dauz electronic drum pads, as an impressive roster of top drummers incorporate Dauz pads into their acoustic drum sets. This is no overnight success story: I first reviewed Dauz pads eight years ago (in the October 1987 **EM**), and company founder Dan Dauz has been building quality electronic drum pads since 1984.

For the past few years, Dauz pads were sold under the KAT name. Now they are available directly from Dauz, either individually or as a complete Dauz Kit system.

YOUR PAD OR MINE?

The Dauz Kit includes ten pads (eight large, round pads and two Bones mini-pads), a heavy-duty chrome rack stand (with all necessary mounting hardware), ratcheted drum key, kick beater, eight ¼-inch cables, Velcro cable ties, drum sticks, and even a Dauz T-shirt! The only things you need to add are a bass-drum pedal, a MIDI sound source, and some kind of pad-to-MIDI converter. (Dauz recommends the popular Alesis D4, which combines a drum-sound source and a converter with twelve pad inputs.)

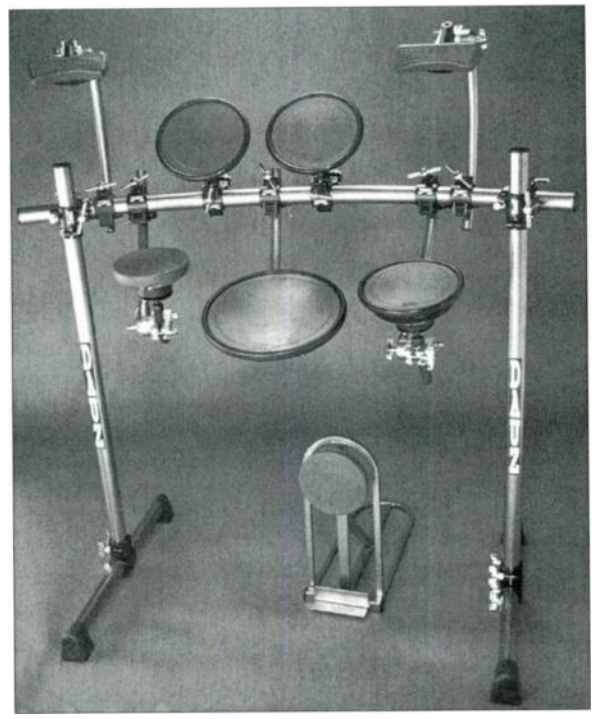
The round Dauz pads are available in 6-, 8-, and 11-inch diameters. The two Bone pads are six inches long and shaped vaguely like dog bones. The Bones' compact size simplifies placement in cramped spaces, and a threaded tube mount lets you mount them over hi-hat stands.

BUILT TO BLAST

The Dauz pads are covered by a thick rubber surface. Inside, a piezo transducer translates the nuances of your performances into analog voltages that are output via recessed ¼-inch jacks. The pad outputs connect to any pad-to-MIDI converter box, such as those made by Alesis, Roland, or Yamaha. I tested the system mostly with an Alesis D4, but it also worked well when connected to the auxiliary pad inputs on my Roland Octapad.

The rubber playing surface extending across the top of the pads is spring loaded, which gives the pads excellent resiliency and a drum-like bounce. In addition, the pads respond equally to velocity strikes anywhere on the playing surface, so there are no "hot spots" (sections that produce excessive output voltage). The piezo transducer is well isolated from the pad mounting system, which eliminates the possibility of false triggering when you're playing adjacent pads.

We drummers have earned a reputation for being tough on equipment, and these pads are built for the long haul. They seem absolutely indestructible. But accidents happen, so Dauz offers a complete line of replacement parts. In fact, the best thing about the Dauz Kit is the availability of a wide variety of extra mounting hardware, pads (priced from $79.95), and accessories, which allow electronic percussionists to customize their kit. I especially recommend the optional 2-zone pad ($169.95), which has separate outputs for the rim and center section. It is ideal for snare/rimshot effects (timbales anyone?) or for simply triggering two sounds from a single hit.



The Dauz Drum Kit includes ten pads with mounting hardware and cables for an excellent price. The piezo pickup is isolated to prevent cross-triggering, and the rubber playing surface is spring loaded, providing a drum-like bounce.

PARTS IS PARTS

The Dauz Kit arrived in several boxes containing all the parts, along with what seemed like miles of stringy plastic wrap. The latter did a fine job of protecting the flawless chrome finish on the rack tubing but took a while to get through. Finally, I had all the parts ready to assemble, but there was no manual or diagram, so it took me an extra ten or fifteen minutes to figure out what went where. It's not really that hard to understand, but the eleven adjustable clamps used in the rack come in five different styles.

Fortunately, once it's set up, you can essentially forget about such trivialities.

It's obvious a lot of thought went into the design of the mounting hardware. The pad-mounting clamps use a split-ring design, so the user merely loosens a wing nut and the clamp removes completely from the rack bars. There's no need to disassemble half the hardware to access a clamp in the center of the system. Four interlocking sleeves align the clamps on the side and top rails. Gigging musicians will appreciate the fact that it's a breeze to precisely reposition the system after packing or unpacking.

The pads are mounted with a unique ball-and-socket arrangement that provides ample flexibility in placing the pads exactly where you want them. This is especially important to users who wish to create tight clusters of pads for percussion-style or melodic playing. Four pads also include long mounting arms, with geared swivel heads, so pads can be placed virtually anywhere within the player's reach.

The kick setup consists of a metal stand incorporating a 6-inch Dauz pad and a mount for a standard bass-drum pedal. Some competing designs can only be used with certain pedals where the beater can be reversed so it moves toward the foot with each stroke. However, the upright design of the Dauz kick accommodates any bass-drum pedal, and it worked well with a Ludwig Speed King, a classic design I've been using for twenty years. To prevent creeping, the underside of the kick stand has rows of tiny plastic "fingers" that grab into carpeting and hold the entire kick setup exactly where you place it, no matter how hard you play.

Product Summary

PRODUCT:
Dauz Drum Kit

PRICE:
$1,499.95

MANUFACTURER:
Dauz Drum Company
338 W. 130th St.
Los Angeles, CA 90061
tel. (310) 366-7301
fax (310) 366-7664

CIRCLE #440 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5
PLAYABILITY	● ● ● ●
VALUE	● ● ● ● ◖

COUNT OFF

Flexible, versatile, and offering real drum feel, the Dauz Kit is an ideal means for drummers and other electronic musicians to explore new performance avenues. The 10-pad system is an excellent value, and even when combined with a pad-to-MIDI converter/sound source, it is priced well below the competition. Besides, it's built like a tank and is a joy to play.

George Petersen *is the editor of* Mix *magazine. As a teenager, he gigged with American rock bands in Italy; today, he writes books, produces records, and occasionally does session drumming.*

Roland DM-800

By Erik Hawkins

This pint-sized DAW is a real knockout.

Although Roland's new DM-800 multitrack hard-disk recorder only weighs twelve pounds, it's no lightweight in the hard-disk recording ring. Drawing on extensive experience in the professional recording market, Roland has pumped up the DM-800 with an impressive array of professional features. These make it a real heavyweight with plenty of punch; despite its size, it's capable of handling almost any recording situation.

The DM-800 is a direct descendant of Roland's DM-80 hard-disk recorder (reviewed in the January 1993 **EM**). A high-quality, if unwieldy, beast that costs more than $10,000 for an 8-track system, the older unit required four separate modules to accomplish what the new machine does in a single, streamlined, 26 × 11-inch chassis.

Although the DM-800 has essentially the same operating system and control surfaces as its predecessor, it has more features and greater DSP power. This enables it to do a few additional tricks, such as pitch correction and time compression.

The unit features an internal power supply and a detachable IEC power cord. The rear-panel power switch is nice and big, so you can grope for it without having to look at the back of the unit.

DISKS AND DACS

Like the DM-80, the DM-800 is a standalone device that does not require a computer. The two Roland DAWs also have the same impeccable specs. Both can record with 16-bit resolution at 32, 44.1, or 48 kHz. The 18-bit, 128× oversampling A/D converters are of very good quality, as are the 18-bit, 8× oversampling DACs. The internal processing is 24 bit.

The unit ships from Japan without internal hard disks. However, Roland Corporation US will install two 2.5-inch, 500 MB hard disks upon request. This gives you about 200 (mono) track-minutes of recording time at 44.1 kHz. Each hard disk handles four tracks (i.e., tracks 1 to 4 are handled by Disk A and tracks 5 to 8 are handled by Disk B). You can copy or move tracks from Disk A to Disk B; the DM-800 copies information between disks in the background on its internal SCSI bus.

Two additional SCSI ports, which parallel Disks A and B, allow more hard disks to be hooked up to a maximum of 4 GB per SCSI chain. This translates to about 24 track-hours of recording time at 48 kHz. Hard disks do not have to be hooked up in pairs, so one 4-track set can have more recording time than the other 4-track set.

The external and internal drives on the two SCSI chains can be organized into three virtual groups, each of which can contain the audio files for up to 50 ongoing projects. A typical disk group might include one or two drives on each SCSI chain—you can't partition individual drives—which would allow you to use all eight tracks on each project. The idea is simply to help you organize your files, and it's especially handy when using removable media.

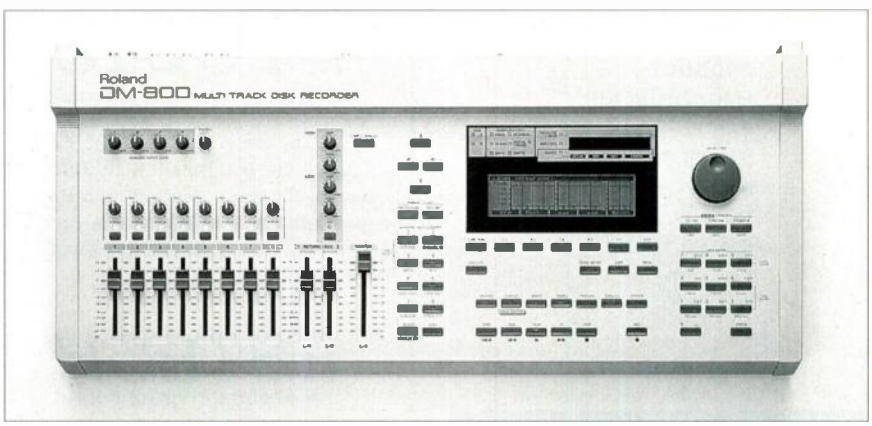
Backup can be done to a variety of media. The rear SCSI connections allow backup to data DAT, or you can copy your projects to external hard disks or most types of removable media (e.g., SyQuest and MO drives). Backup to standard audio DAT is accomplished with the DM-800's S/PDIF output. You can back up one project or all projects at the same time, except with audio DAT, where backup can only be done project by project.

AUDIO CONNECTIONS

Although the DM-800 is an 8-track machine, there are only six audio inputs. Each of the four balanced (TRS), analog ¼-inch inputs on the rear panel has a 20 dB variable-gain control on the unit's face for matching +4 to -10 dBm levels. A stereo coaxial S/PDIF digital input is available simultaneously with the four analog inputs, which brings the total number of inputs to six. Each of these inputs can be assigned to any or all tracks or mixer channels.

The eight simultaneously available audio outputs aren't configured in the usual way. Four are balanced (TRS), analog ¼-inch outputs, and the other four are in the form of two stereo coaxial S/PDIF digital outputs. Any or all of the eight tracks can be assigned to any or all outputs. The headphone output is on the unit's rear panel, and a headphone volume knob is easily accessible on the face.

Roland's Multipurpose Digital Bus (RMDB) is an 8-channel digital-audio connector that allows the DM-800 to



Roland's DM-800 multitrack hard-disk recorder is a boon for studios where space is at a premium and for recordists who are on the go.

interface with other digital audio devices. An optional DIF-800 interface ($1,095) is available for the Alesis ADAT, TASCAM DA-88, and Sony-protocol video editors via RS-422.

I understand that Roland was trying to keep the cost of the DM-800 down, but it certainly would have been nice to have a full set of balanced ¼-inch inputs and outputs. The way the unit is currently set up necessitates keeping a couple of extra DACs and ADCs handy if you want eight channels of analog I/O, which is impractical for many personal studios. Fortunately, the Roland DA-400 ($646) provides four channels of D-to-A conversion at a reasonable price.

USER INTERFACE

The face of the DM-800 is covered with things to push, pull, and turn. It is pleasingly tactile and logically laid out considering the unit's capabilities. The front panel is divided into mixer, locate, command (e.g., transport controls and mode select keys), and data-entry sections. Each section is easily discernible thanks to the way the controls are grouped, and there are plenty of dedicated buttons.

▼

The DM-800 is pleasingly tactile and logically laid out.

The main display window is a nice, big LCD that shows system, track, and waveform parameters. Data entry can be accomplished by three different means: single increment/decrement buttons, a standard 10-key pad, or a rotary alpha dial. In addition, a big, easy-to-read, 9-digit alphanumeric LED shows the project's current location. It can display SMPTE time (with subframe accuracy), minutes/seconds, or measures/beats/clocks.

Basic transport functions are accomplished with the usual Play, Record, Rewind, Fast Forward, and Stop buttons. As with most hard-disk recorders, the Stop button doubles as a Pause button. The Rewind and Fast Forward buttons are a bit aggravating because they don't lock in with a single press, as

you'd expect. Instead, one button-press advances or rewinds the material by a single second. Holding the buttons down results in painfully slow movement that does not increase in speed the longer the buttons are depressed. Fortunately, the Play and Record buttons work in the conventional way.

A few nice extras also are provided. For example, the DM-800 supports MIDI Machine Control (which I was unable to test), and Record and punch in/out can be triggered via MIDI.

BASIC OPERATION

There are seven modes of operation: Record, Playlist, Mixer, Tempo, Trigger, Catalog, and System. Each can be accessed with a dedicated button. Multiple windows are available in each mode, which can be called up with a set of five "soft" keys located directly beneath the LCD.

The names of the modes are self-explanatory, except Trigger Play mode, which allows an unlimited number of sequential sounds to be linked to a MIDI trigger or note (see **Fig. 1**). This is perfect for dropping in Foley effects on the fly, but it doesn't pitch-shift the sounds and spread them across the keyboard like a sampler does.

When the unit is in Play you must press Stop to enter another mode. This is a bit annoying when you are in the Playlist mode and get inspired to record a track or an automated mixer move using the Record or Mixer modes. It would make the unit's operation much more fluid if you could move between these modes while the unit is in Play.

There are a total of 40 locate points, which Roland calls Markers. However, they are divided into five Banks, making only eight Markers available at a time. The eight available Markers have their own dedicated buttons, but I was surprised to find that you can't name the Markers or the Marker Banks.

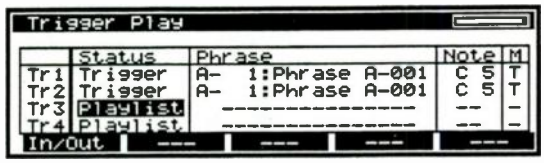


FIG. 1: Trigger Play mode lets you use MIDI notes to trigger an unlimited number of sequential sounds, which is great for dropping Foley effects on the fly. Front-panel Keypad buttons 1 to 8 also can be used to trigger the Phrase on the corresponding track. (Courtesy Roland Corporation US.)

In addition to the Markers, there is a Phrase Locate function that allows you to jump to the beginning or the end of a recorded take. This is a nice feature for finding the areas of a track that have been recorded on but not marked. A Jump function allows you to type in a SMPTE time and jump to that exact location. The Scrub function helps you find really tight cues, but unfortunately, the system I tested only let me hear one track at a time. According to the manufacturer, this was due to a minor bug that will be corrected in version 1.24.

Another handy feature is Section Loop, which constantly repeats the selected audio from the beginning to the end of the loop, letting you record overdubs sequencer-style. If you trigger the function while in the middle of the selection, the DM-800 waits until the loop starts over and then kicks in. Every take is saved. If you think you'll use this feature often, make sure you have the latest version of the DM-800 OS, as Section Loop had a minor bug in the early versions. (I reviewed version 1.21.)

EXTERNAL EQUIPMENT

Like many Roland samplers (such as the S-550 and S-770), the DM-800 has a video display output that supplements the LCD screen. But unlike the samplers, the DM-800 has an wide selection of video outputs, including digital RGB, S-VHS, and standard RCA composite video. All three are switchable between NTSC or PAL.

The bottom third of the video display is an exact duplicate of what you see on the LCD screen. The middle portion shows you all the tracks and their status. The recorded area of the tracks is indicated by rectangular blocks. The top third of the display is split between a meter bridge and a counter that displays SMPTE time (minutes/seconds) and measures/beats/clocks. (There are 250 clocks per quarter note.) The display's colors are programmable and should accommodate a wide assortment of tastes (see **Fig. 1**).

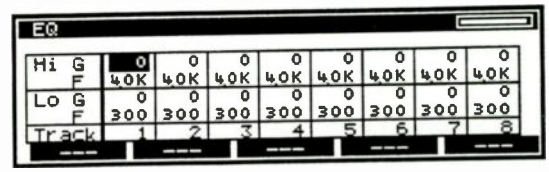


FIG. 2: The DM-800's 2-band channel EQ is semiparametric, meaning you can control the center frequency (F) and the amount of gain (G) but not the bandwidth. In this example, all eight tracks have the high band set at 4 kHz and the low band at 300 Hz. (Courtesy Roland Corporation US.)

The video display is invaluable; I hated using the machine without having it hooked up. However, working with the video display felt uncomfortably like working on an early DOS computer in CGA mode. It looks like an old Apple II display, and there's no mouse support, just a set of cursor keys for moving up/down and left/right, which is far less convenient. (The S-550 and S-770 samplers let you use a mouse; why not the DM-800?)

On the other hand, unlike the S-550 and the S-770, the DM-800 has an ASCII keyboard input. Plugging in any PC AT keyboard lets you duplicate the unit's front-panel controls with keystroke commands. This is a great feature, because it creates a comprehensive remote controller that is the cheapest I've ever run across for this type of device.

There is also a footswitch jack on the back of the DM-800 that accepts a momentary switch. It can be used to punch in/out of Record or to enter a tap tempo when creating a tempo map.

SYNCHRONIZATION

Roland's long history of making synchronization units shows in the DM-800. It reads and generates both MIDI Time Code and SMPTE, with support for 24, 25, 29.97, and 30 fps SMPTE rates, in addition to 29.97 drop-frame. An error-level adjustment control allows the unit to ignore SMPTE code with high error rates. Moreover, the DM-800's internal word clock can be set to recognize external digital or SMPTE clocks in order to control the exact sample playback rate. The unit can also reshape incoming SMPTE code and pass it to the output.

I locked the DM-800 up with a variety of software and hardware, including Mark of the Unicorn's *Performer* sequencer, Opcode's *Studio Vision* digital audio sequencer, and the Alesis AI-2 and BRC. In every instance, it synched marvelously. Just for fun, I also striped an

ADAT tape with 29.97 nondrop SMPTE and fed it to the DM-800 while simultaneously changing the pitch on the ADAT. Amazingly, the DM-800 tracked the entire range of pitch changes accurately, without missing a beat.

My only complaint about the synchronization implementation is that the SMPTE offset is not comprehensive, only allowing offset times to be set in 10-minute segments. Luckily, you can move recorded material in time on the tracks and achieve offset times with subframe accuracy. This method works, but it is not what most people are used to when setting offset times.

MIXING AND EDITING

The DM-800's mixer is impressive. There are eight main faders, which are associated with the eight audio tracks, and two stereo aux returns, for a total of twelve channels. The faders feel nice and solid.

▼

Be prepared for a steep learning curve.

All mixer functions can be automated, including levels; panning; mutes; and 2-band, semiparametric EQ (i.e., with cut/boost and center-frequency controls, but not bandwidth; see **Fig. 2**). Flexible internal digital routing is available for everything from internal mixdowns to merging multiple tracks to a single output. Having what amounts to an internal digital patch bay is a real plus. The only things missing are a solo function and mic inputs with phantom power (which would have been great in a portable unit).

Pretty much every edit you can imagine can be accomplished, including splicing and moving, crossfades, time compression/expansion, pitch correction, and finding zero crossings. However, don't plan on getting the job done quickly and painlessly unless it's something really simple (e.g., deleting a take). Some of the higher-level DSP functions, such as time compression/expansion and pitch correction, take quite a while to process (Roland's representative could not quote the unit's exact processing speed for these DSP

functions, but I suspect it's pretty slow.)

Processing time is inversely proportional to audio quality, so Roland lets you choose between three processing settings (High, Mid, and Low). The High setting takes the longest time to process but yields the best audio resolution, while Low is much faster but yields the poorest resolution.

At first, I felt like I was banging my head against a brick wall when trying to edit on the DM-800. After a while, though, I started getting the hang of things, and the moves became easier. If you are used to computer-based DAWs and you plan on editing with this unit, be prepared for a steep learning curve, especially if you're not used to Roland's user-interface concepts.

When learning how to get around on the DM-800, especially in Edit mode, you will inevitably make mistakes. As with any hard-disk recorder, you should save to disk as soon as you get a take, edit, or mix you like. This is especially important with the DM-800, however, because it doesn't have an Undo (last command) function. Fortunately, it has a Revert (to the most recently saved version) function, so if you make a mistake but you saved recently, you shouldn't lose much work. Still, an Undo function would have been greatly appreciated.

THE BOTTOM LINE

I usually judge a piece of gear on three main things: bang for the buck, sound quality, and ease of use. The DM-800 is a bit pricey, but it's still a good deal, especially when you compare it to the DM-80 and consider that you don't have to buy a computer to run it. True, fast '486 PCs can be had for a song (and Pentiums and Power Macs for a short medley), so you might be able to outfit yourself with a more powerful, computer-based DAW for a comparable price. On the other hand, you can sink a fortune into a top-flight computer-based system (e.g., Digidesign's Pro Tools), especially when you start adding up the extras.

Besides, you can't tuck a desktop computer and external fader box under your arm and carry it with ease to an outside studio or live recording gig. With the DM-800, you can make the scene with your back intact, and you'll be ready to record immediately.

I love the way Roland equipment sounds, and this unit is no exception. The DM-800 has Roland's classic fat, warm sound, much like that of the S-770. According to Roland's representative, it is impossible to tell the difference between the output of the DM-80 and the DM-800. From what I heard, I agree.

How user friendly is it? This is a more difficult call. The unit does nearly everything I wanted it to do, but I had to spend a lot of time guessing at how to do things because the DM-800's operating system is not intuitive. For example, several functions have the same name but do different things depending on what mode you are in. If you are used to Roland equipment, you have a distinct advantage but not a free ride.

The manual didn't help much, especially with its abbreviated index. It seemed more oriented toward explaining what each function does, rather than how you can use them in each mode. Fortunately, by the time you read this, the unit will ship with a 3-hour, application-oriented instructional video, which should help greatly.

The bottom line is that the DM-800 is an extremely powerful piece of gear with a lot of good qualities. Its portability is a major plus for users who often work on the go, such as independent producers. If you are shopping for a portable DAW and you can afford the price of admission, the DM-800 merits serious consideration.

Erik Hawkins *is a producer and musician in Los Angeles County and the San Francisco Bay Area. He's about to go off the musical deep end.*

Product Summary

PRODUCT:
DM-800 hard-disk recorder
PRICE:
$6,295 (without hard disk)
MANUFACTURER:
Roland Corporation US
7200 Dominion Circle
Los Angeles, CA 90040-3647
tel. (213) 685-5141
fax (213) 722-0911
CIRCLE #441 ON READER SERVICE CARD

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

Steinberg ReCycle! 1.1 (Mac)

By Eric Leach

Shift sampled grooves with this innovative software.

Don't let the name of this program fool you. At first glance, you might think it's designed to help you recycle your old bottles and newspapers. Alternatively, you could get the impression that this is a rehashed version of some older software. However, Steinberg's *ReCycle!* is unlike any application I've seen. (The program is available for the Mac and Windows; I reviewed the Mac version.)

So what is it, exactly? As with anything new and different, it's hard to classify *ReCycle!* in a neat, descriptive category. Steinberg suggests it can be called a "breakbeat analyzer and editor." I tend to describe it as a specialized tool for working with sampled grooves.

Basically, *ReCycle!* analyzes sampled grooves and breaks them into tiny pieces that can be manipulated in a manner similar to working with grooves of individual sounds. Sound complicated? It's not. The potential and simplicity of *ReCycle!* will become evident as we discuss the program's abilities in greater detail.

In addition to the basic computer requirements, you need a sampler and a sequencer that can work with Standard MIDI Files (SMFs) to fully exploit *ReCycle!*'s potential. Currently, *ReCycle!* supports Akai samplers (S1000 or later) and Digidesign's SampleCell, SampleCell II, and SampleCell PC. The next version (1.5), which should be shipping this fall, will also support the Roland S-760; Ensoniq EPS, EPS-16 Plus, and ASR-10; and Kurzweil K2000 and K2500. The Mac version supports AIFF, Sound Designer, and Sound Designer II files; the Windows version supports WAV and AIFF files.

LOAD 'EM UP

To work with a digital audio file, *ReCycle!* must first load it into RAM, so the more available RAM you have, the more and longer sound files you are able to load.

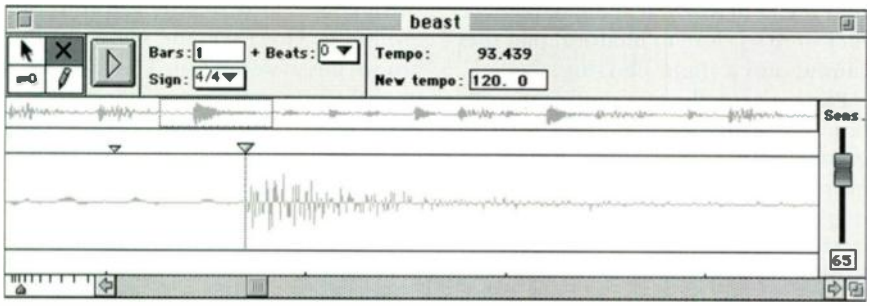


FIG. 1: *ReCycle!*'s main window provides access to all functions. The thin Overview window displays the entire audio file while the main window below displays any portion you wish.

Although this may seem like a disadvantage at first, it allows *ReCycle!* to work quickly. In addition, this approach makes sense because sampled grooves are typically short in length. If your grooves are stored in your sampler, *ReCycle!* can retrieve them via SCSI or MIDI Sample Dump.

One of the few annoying aspects of *ReCycle!* is the fact that it can load only 16-bit, 44.1 kHz files. All other file types must be converted before use. Furthermore, it can load only mono files. (Version 1.5 will accommodate stereo files.) If your file contains more than one audio channel, *ReCycle!* asks you which channel you want to load.

BREAK 'EM DOWN

All of *ReCycle!*'s functions are available in one main window, which appears after an audio file is loaded (see **Fig. 1**). All or part of the waveform is displayed in the main working area, which includes zoom controls to zero in on the desired portion. Above the working area, an Overview window displays the entire waveform, identifying the portion in the main window with a "thumbnail" viewer. Steinberg fans will instantly recognize this simple method for viewing and working on waveforms.

A Play button lets you hear the file, and a volume control in the Options menu can be used to balance the sound from *ReCycle!* with the sound from other applications. The most important control is the Sensitivity fader at the far right of the main window, which works like the threshold control on a compressor. As you raise this fader, you see a number of lines appear across the waveform (see **Fig. 2**). These represent *ReCycle!*'s "slice" points at which the file will be divided into a series of smaller files. For instance, a drum groove is divided into a series of individual audio files, each representing the successive drum hits in the groove.

ReCycle! locates amplitude peaks to define these slice points and does it exceptionally well. However, you can also add slices manually or "hide" unwanted slices. When the cursor is placed over any slice, it becomes a speaker icon that lets you audition each slice individually. I got the best results by setting the Sensitivity fader a little too high and hiding unwanted slices. At the simplest level, you can use *ReCycle!* in this way to automate the tedious process of finding the right loop points

Product Summary

PRODUCT:
ReCycle! 1.1 sample editor

PRICE:
$199

SYSTEM REQUIREMENTS:
Macintosh: SE/30 or later with 4 MB RAM; System 6.0.7 or later; hard disk. Apple Sound Manager 3.0 or later recommended. Windows: 80386 or better with 4 MB RAM (8 MB recommended); Windows 3.1 or later; MME-compatible sound and MIDI card. Adaptec ASPI-compatible SCSI card recommended.

MANUFACTURER:
Steinberg North America
9312 Deering Ave.
Chatsworth, CA 91311
tel. (818) 993-4091
fax (818) 701-7452

CIRCLE #442 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ● ● ◖
EASE OF USE	● ● ● ● ●
DOCUMENTATION	● ● ● ●
VALUE	● ● ● ● ◖

for your samples or extracting individual sounds from long samples.

SLICES, DICES, JULIENNES

After the groove is properly sliced, you can begin to explore the program's creative possibilities. Beneath the main waveform display are left and right Locators (see **Fig. 2**), which let you set up the playback loop or the portion of the waveform you want to use as your groove. The Locators snap to the nearest slice point, allowing you to instantly define the groove.

At this point, you can tell *ReCycle!* the length of your groove in bars and beats with a time signature, after which it automatically (and accurately) calculates the groove's tempo. This feature is excellent for groove-oriented musicians because you can build a sequence around a sampled groove without the hit-or-miss guesswork of manually setting tempos. Although *ReCycle!* offers relatively few time-signature options, I was able to circumvent this limitation by adding beats to a standard time signature (e.g., one 4/4 bar plus one beat for one 5/4 bar).

HERE'S A SAMPLE

After the aforementioned manipulations, the groove can be transmitted to your sampler as a set of perfectly sliced rhythmic components. *ReCycle!* is smart enough to place these files logically within your particular sampler. For instance, the files appear as a Program in the Akai samplers with each slice assigned to successive note numbers.

ReCycle! can also create a Standard MIDI File containing these note numbers at rhythmic locations based on the relative timing of the groove slices. When you send this SMF to your sampler, the groove is perfectly re-created. When I compared this SMF playback to the original audio file using adjacent tracks in a digital audio sequencer, they were so close to identical that they caused only a slight flanging.

I encountered one significant problem while preparing to export my groove to an Akai S1000PB. *ReCycle!* requires as much additional memory for transmission as it uses to hold the slices in the first place. If you're not careful to allocate memory ahead of time, you can't transmit. In this case, your only option is to quit *ReCycle!*, allocate more memory, and re-create the slices. You can't even save your work as a *ReCycle!* file and open it after allocating more memory; the program only saves the slices as a series of separate AIFF or Sound Designer files (AIFF or WAV in the Windows version). You can save the portion between the left and right Locators as a single file if you eliminate the slices, which is great for people who are looking for the perfect loop.

▼ **The Stretch feature lets you slow down the tempo without unwanted silence between slices.**

LET'S GROOVE!

I tried *ReCycle!* in a series of applications. First, I decided to build a sequence around my favorite sampled drum groove. I created an SMF from the groove and began sequencing around it. However, the tempo of the groove was a bit too fast. To solve this, I used the program's unique Stretch feature. By adding a natural-sounding decay tail to each slice, this feature lets you slow down the tempo without any unwanted silence between slices. Even drastic tempo reductions worked nearly perfectly.

Other manipulations of the groove from within the sequencer also worked well. By working with the notes that trigger the slices, I was able to change the feel of the groove and even quantize it. I also created a groove-quantize template that I applied to other sequenced drum parts. Imagine being able to apply the feel of your favorite drum licks to your sequenced drums!

Altering the groove within the sampler also provides a new realm of possibilities. I replaced my kick samples with other sounds, sent the snare hits to individual outputs for effects processing, and changed the level of the hihats, all without changing the feel of the groove. A warning, however: this kind of editing requires a groove with fairly distinct rhythmic components and possibly some trial-and-error slicing within *ReCycle!*.

Although drum grooves are the easiest to manipulate, it is also possible to use pitched grooves, such as guitar riffs, for many applications. I successfully built a sequence around a guitar riff using various tempos and even changed the pitch of the riff into various keys without affecting the feel.

CONCLUSIONS

The key to working successfully with *ReCycle!* is selecting grooves with discernible attacks for each sound. Although *ReCycle!* is especially good at analyzing and slicing audio files, it may have trouble with samples that are smeared or not recorded well.

I've touched on several applications, but these are only the tip of the iceberg. Suffice to say that *ReCycle!* does all the things it's supposed to do, and it does them very well. This program should be on the short list of anyone working with sampled grooves.

(Thanks to Betalab, Giles Reaves, and Merrick Music.)

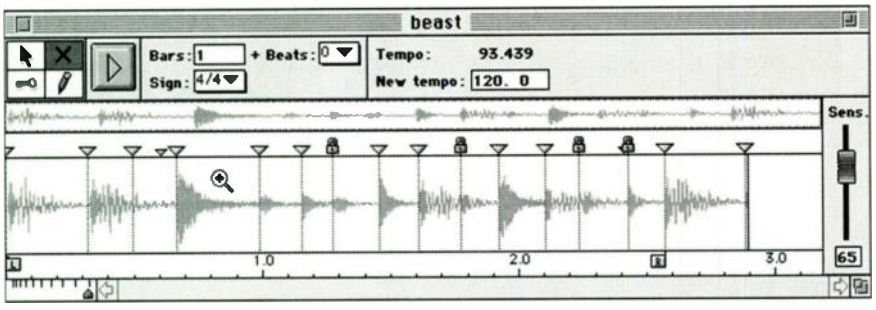


FIG. 2: The audio file is sliced into small chunks that start at amplitude peaks. *ReCycle!*'s Sensitivity slider determines how large these peaks must be to serve as slice boundaries.

Eric Leach *is a devoted husband and a wannabe NBA-draft analyst.*

dbx 290 Stereo Reverb

By Peter Freeman

A venerable dynamics-processor company enters the reverb market.

Although dbx has a well-established reputation in the world of dynamics processors, the company is only now making its way into the area of digital reverbs. The company's initial offering in this market, the 290 stereo reverb, is clearly intended for the budget-conscious user, offering an alternative to the Alesis MidiVerb and Lexicon LXP-1.

FIRST GLANCE

The 290 comes in a standard 1U rackmount case with the expected (if not welcome) wall-wart AC supply. The back panel includes ¼-inch jacks for the left and right inputs and outputs, a bypass-footswitch jack, and a MIDI In port (but no Thru). The front-panel controls are straightforward: three groups of LED-lit buttons provide control over reverb type, Size/Gate Shape, and Color.

The available reverb types are Room, Hall, Chamber, Plate, Cathedral, and Gated. These algorithms are modified by the Small, Medium, and Large Size buttons, which simulate spaces of different size in all but the gated algorithms, as well as the Dark, Medium, and Bright Color buttons, which modify the overall timbre of the processed sound. The Gate Shape buttons affect the envelope of the Gated algorithm. There are only four other controls on the unit, all of which are knobs: Decay, Wet/Dry Mix, Input, and Output. A pair of 4-LED bar-graph meters indicate the input-signal level.

Unlike many budget reverbs, the 290 is a discrete stereo device, i.e., it can process the left and right inputs independently, rather than summing the inputs, processing the mixed signal, and simulating a stereo output. It offers three modes of operation: Stereo In/Out, Mono In/Stereo Out, and Mono In/Out. The 290 automatically selects the appropriate mode depending on which inputs and outputs are connected.

There are no user-memory locations, and you cannot call up factory presets from the front panel. Instead, there are 54 different combinations of front-panel settings, which comprise all possible processing options. However, these combinations *are* available as presets that can be called up with MIDI Program Change messages. It seems odd that you can select presets from MIDI but not from the front panel. Thanks to the programming simplicity of the unit, though, it's not a problem.

You can also change Reverb Decay time with Control Change 01 (Modulation) messages over a range of twelve values. As limited as this is, it offers a degree of automated control during mixing.

SOUND QUALITY

As soon as I received the review unit, I patched it into the aux sends and returns of one of the Mackie CR-1604 mixers that live in my pre-production studio. I experimented with the 290's various settings on drums, synths, miscellaneous samples, and vocals.

The first thing I noticed was that the algorithms in this unit have a somewhat artificial quality, which tends to manifest itself as a perceived lack of depth and richness in the sound of the reverb. In addition, the Color choices tend to be quite drastic: the Dark setting is almost too muffled, and the Bright setting produces somewhat harsh results. I ended up using the Medium Color setting more often than the others.

The Cathedral and Chamber algorithms were the most useful overall, but the Gated and Plate settings weren't flattering on most of the material I used. This is important because

The dbx 290's overall sound is unimpressive, but the low-cost, easy-to-use unit is acceptable as an extra processor for the home studio.

dbx 290 Specifications

Frequency Response	20 Hz to 20 kHz (+0/-3 dB)
Sampling Rate	40 kHz
ADC/DAC Resolution	18 bits
Internal Resolution	24 bits
THD	<0.02% @ 1 kHz
S/N Ratio	>90 dB (max signal @ 22 kHz bandwidth)

the 290 is intended to minimize programming and knob-twiddling with a selection of ready-to-go, useful sounds. Unfortunately, this effort is not entirely successful in this case. If the sound isn't right, there isn't much you can do to change it.

By itself, the 290 seemed to work best on softer synth and guitar sounds for which a wash was required. Drums are always an acid test for a digital reverb, and they tended to reveal the 290's shortcomings (i.e., ringy and rather thin-sounding reverb). I achieved the best results when I used the 290 in conjunction with another processor. Feeding the same signal to two different reverb units produced some nice sounds.

CONCLUSION

Personally, I prefer the sound of the Alesis MicroVerb and MidiVerb series to that of the 290. These days, any budget-priced reverb must meet or exceed this standard, and I didn't think the 290 quite made it.

Product Summary

PRODUCT:
290 stereo reverb
PRICE:
$379
MANUFACTURER:
dbx
8760 S. Sandy Parkway
Sandy, Utah 84070
tel. (801) 566-8800
fax (801) 566-7005
CIRCLE #443 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5
FEATURES	● ● ◖
EASE OF USE	● ● ● ●
SOUND QUALITY	● ● ◖
VALUE	● ● ●

Nevertheless, if you are looking for an inexpensive and simple-to-use reverb for a home/demo studio where high-end reverbs aren't in the budget, the 290 is worth considering. It will certainly function well as a second or third reverb to add an extra dimension to a mix without tying up your main reverb. With careful adjustment of the 290's few parameters, satisfactory results are possible on many different instrument sources.

Peter Freeman *is a freelance bassist/synthesist and composer living in New York City. He has worked with such artists as John Cale, Jon Hassell, Chris Spedding, L. Shankar, Sussan Deihim, Richard Horowitz, and Seal.*

Dittamo Ditto Discs/ KAT Kits

By Dan Phillips

Acoustic percussion samples for discriminating tastes.

For some years now, KAT has been a major manufacturer of MIDI percussion hardware, including the drumKAT, malletKAT, and trapKAT controllers. More recently, the company has started to distribute sampling CDs and CD-ROMs of drum sounds, including Jay Dittamo's *Ditto Discs* CD-ROM. When I discovered KAT was also working on their own CD-ROM of drum sounds, *KAT Kits Vol. 1*, I was eager to compare the two collections.

The Dittamo and KAT CD-ROMs are available in Akai S1000 format only. Fortunately, Akai-format samples can be imported into many other samplers, though you might have to do a little tweaking in some cases.

KOOL KAT KITS

Created by KAT's Mario DeCiutiis and William Stahmann III, *KAT Kits Vol. 1* contains twelve full drum kits—four jazz kits and eight rock kits—along with extra snares, kicks, hi-hats, and ride cymbals. All programming was done on an Akai S3000 sampler. Five different drum kits were used, most of which appear multiple times with different tunings or amounts of ambience. These include a DW Custom Maple and a DW Custom Spruce, a 1946 Slingerland Radio King, a 1949 Gretsch Round Badge, and a 1992 Johnny C.

All of the kit samples are mapped according to the General MIDI specification. There are two different versions each of the kick and snare, snare sidestick and rim shot, hi-hats (open, closed, footpedal, and foot splash), four toms, two crashes, splash and china cymbals, ride and ride bell, cowbell, woodblock, and triangle.

Unfortunately, the full kits are the only programs provided. It would have been convenient to have each sound available as a separate program, as this would ease mixing and matching between different kits.

CROSSFADE TO EXPRESSION

KAT is an enterprise by and for drummers, and the programming of the *KAT Kits* really shows that. Each kick, snare, and tom uses four different samples at varying strike velocities, switching and crossfading between them to allow truly expressive playing. (The snare sidestick and rim shot generally use only a single sample apiece, however.)

The hi-hat samples are almost as exhaustive, with either three or four samples for closed hits and two for open hits. All crashes and splash cymbals are sampled with soft and hard attacks, and some of the rides are multisampled, as well. There are even separate samples for "holds" of each crash, a performance technique in which the cymbal is struck and then muted by grabbing it with your hand.

In addition to the hi-hats in the kits, there are two sets of hi-hat samples designed for a special feature of the drumKAT and trapKAT that allows a continuous hi-hat pedal to select eight different degrees of hi-hat openness, along with foot chick and splash. It's a pity that only a single Velocity level is included for each sample.

All of this makes these samples very satisfying to play. I loaded them up, plugged in my drumKAT, and wailed away for hours. When I played lightly, it felt like I was just barely touching the skins with the sticks, and when I played hard, it really seemed like I was digging into the heads.

DOWN TO SOUNDS

KAT's focus for *KAT Kits* is on natural-sounding timbres, often with "vintage"

KAT, Inc.'s *KAT Kits* sample CD-ROM offers eight rock kits and four jazz kits, along with a selection of individual drums and cymbals. The emphasis is on natural-sounding timbres, rather than heavily processed sounds.

character. There are a couple of deep, tight pop kicks, but there also are blatty, boomy, and flabby ones in which you can almost see the heads shake back and forth. The snare drums aren't gated or exaggerated; many have been set up to highlight the rich buzz of their loose snares (the wires under the drum that produce the "snappy" sound).

Jazz drummers, or producers looking for authentic-sounding jazz drum samples, will certainly appreciate this collection. For instance, there are beautiful samples of the 1946 Slingerland Krupa set with original calf drum heads, and you get a choice between a big, boomy kick or one that is, by modern standards, almost laughably light and small.

The cymbals and hats are standouts, superb and well-suited to general-purpose use; they would be a fine addition to anyone's drum-sound collection. Many of the toms could have been allowed to ring longer, but this doesn't seem to stick out in context, and the 4-way multisampling really pulls its weight during fills. Many of the snares would fit in well on rock or alternative tracks when you're looking for a distinctive drum sound.

Ambience is the one area in which the *KAT Kits* have problems. In order to fit these already memory-heavy kits into a 16 MB sampler, all of the samples are monaural. I don't mind this for dry samples, as outboard reverb and panning can provide sufficient stereo imaging. For ambient sounds, however, it's problematic, and for sounds with long reverb tails, it's pretty glaring.

Several of the kits (Rock 5, 6, and 8) specifically feature wet, reverberant samples, but to my ears, the fact that they are in mono greatly reduces their usefulness. There's a considerable amount of empty space left on the disc, as the sounds take up about 200 megabytes; I wish KAT had used some of the disc's remaining capacity to include the stereo versions of these files for those with enough sampler memory to handle them.

DYNAMIC DITTO DISCS

The *Ditto Discs* CD-ROM takes a slightly less in-depth look at a lot more drums. The library includes 25 full kits, along with several banks of Latin and orchestral percussion, timpani, vibraphone, marimba, xylophone, bells, and various synth drums.

As with the *KAT Kits* CD-ROM, the drums in *Ditto Discs* are arranged in General MIDI fashion. Each kit includes kick; snare; sidestick; between three and five toms; open, closed, and foot-controlled hi-hats; two crashes; splash, china, and ride cymbals; and cowbell. Some of the kits also include extra hi-hat samples.

Product Summary

PRODUCT:
Dittamo *Ditto Discs*
sample CD-ROM
PRICE:
$219
DISTRIBUTOR:
KAT, Inc.
53 First Ave.
Chicopee, MA 01020
tel. (413) 594-7466
fax (413) 592-7987
e-mail kat1993@aol.com
CIRCLE #444 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5
AUDIO QUALITY	● ● ●
DOCUMENTATION	● ● ● ●
VALUE	● ● ●

Jay Dittamo's *Ditto Discs* CD-ROM includes 25 sampled kits, along with several banks of assorted percussion and synth drums. The snares are especially excellent.

You'll find far fewer multisamples than in the *KAT Kits,* but all of the snares offer four Velocity switches, and the hi-hats use two. Some percussion sounds make effective use of Velocity switching, and the rides offer the traditional edge/bell switch, but the rest of the drum sounds are generally captured with only a single sample. As with the *KAT Kits* collection, all sounds are monaural, but this set avoids problems with ambience by keeping the samples fairly dry.

The snares are the best part of this collection and may be worth the purchase price on their own. They are expressive, natural sounding, and almost processing free, allowing you to create your own timbral variations using outboard compression, equalization, and reverb. This means you need to dedicate a sampler output to the snare drum, so if your system only has stereo outputs, the sounds might be more difficult to work with. If you are in the habit of using external processing already (as I am), they should fit right in.

Most of the kick drums consist of two samples layered together; presumably, these are samples of the same drum recorded with different microphones. While this is a traditional recording technique, I think it's probably more practical to create single samples using a blend of the mics. At any rate, I found this approach less than effective sonically; most of the kicks seemed to improve by choosing one version and removing the other.

Many of the toms have really pleasant, round tones, but the lack of Velocity switching really hurts them. The hi-hats are decent, and I was pleased with the cymbals before I heard the better ones on the KAT disk. A little expressiveness would have gone a long way, here.

The percussion sounds are pretty good, with lots of variations and scores of different instruments represented. Most of these make effective use of Velocity switching either to shift dynamics or to select between different performance techniques (such as the bongo samples, which switch between muted and open hits).

Product Summary

PRODUCT:
KAT Kits sample CD-ROM
PRICE:
$219
MANUFACTURER:
KAT, Inc.
53 First Ave.
Chicopee, MA 01020
tel. (413) 594-7466
fax (413) 592-7987
e-mail kat1993@aol.com
CIRCLE #445 ON READER SERVICE CARD

EM METERS	RATING PRODUCTS FROM 1 TO 5
AUDIO QUALITY	● ● ●
DOCUMENTATION	● ●
VALUE	● ● ●

DRUMMING UP BUSINESS

Both the Dittamo and KAT discs emphasize natural, raw sounds, as opposed to slickly produced pop drums. This means that they won't be for everyone. For jazz drummers, I can wholeheartedly recommend *KAT Kits Vol. 1;* it offers several useful, natural-sounding jazz kits that have exceptional expressiveness.

Those looking for their first sample CD of pop and rock drum sounds probably should look elsewhere. For example, Bob Clearmountain's *Drums II* and Ross Garfield's *Drum Doctor Does Drums* feature more "produced" timbres and would be better starting places for these genres.

However, if your basics are covered and you're looking for new inspiration for the percussive muse, these two discs have much to offer. In addition to the jazz drums, *KAT Kits Vol. 1* has great cymbals and hats, along with many interesting snares and toms. And I wasn't kidding when I suggested that the snares alone might justify the price of *Ditto Discs;* these dry, multisampled, expressive snares are exactly what I've been looking for recently. Whenever I can say that, I'm happy.

(Thanks to David Vogel and Danny Fouche of Gelb Music.)

Dan Phillips *is a principal in Touch Productions, providing music for film, television, and album projects. He happily spends his weekdays as part of the team at Korg Research and Development.*

Sabine FBX-1802

By Lori Bolender and Mike Cutter

How to kill feedback without even trying.

We've changed our tune. When we reviewed the Sabine FBX-900 Feedback Exterminator (see the March 1993 **EM**), one of the company's early, single-channel units, we had mixed feelings about it. The concept was nice, but the device had a few inherent problems that limited its flexibility.

When we pulled the Sabine FBX-1802 Dual Feedback Exterminator from its packing material, our first impression was "same thing, different box." But we were mistaken. The FBX-1802 offers improved audio quality, faster response, and better shielding and is far more flexible than its predecessor. It also incorporates several features that enhance the unit's usefulness in professional applications.

The dual-channel device was not designed for "voicing" or tailoring a system to a certain sound but for automatically controlling feedback. It can be inserted in a number of places within

the audio signal chain and makes an excellent substitute anywhere you would normally put a ⅓-octave or parametric equalizer to control feedback. It can be much more effective than a standard equalizer in this role because it can detect and respond to feedback much faster than a sound engineer who already has his or her hands full. In addition, thanks to its narrow filters, the unit does not color the sound as much as wider-bandwidth equalizers.

BASIC FEATURES

Those narrow filters are the heart of the FBX series. Each of the FBX-1802's nine digital notch filters per channel is capable of an astounding 40 dB of attenuation at the designated frequency band (between 20 Hz and 20 kHz). When the FBX senses feedback in the signal path, its 32-bit, floating-point, digital signal processor automatically inserts a user-selectable, 0.1- or 0.2-octave filter with only the necessary amount of cut to eliminate the feedback. Sabine claims the unit's response time is approximately 0.4 seconds at 1 kHz, which is plenty fast enough to squelch feedback.

The single-rackspace device incorporates both balanced XLR and ¼-inch tip-ring-sleeve connectors, which also accept unbalanced sources. The rear panel features pin 1, chassis ground-lift switches for the inputs and outputs to assist in troubleshooting hums and ground-loops. The 20V outboard power supply includes a mounting hole, which simplifies hanging the cumbersome transformer to the side of your rack without hogging two or three outlets on your power strip.

Each channel has a rotary knob for adjusting the clip level. The knob maintains unity gain no matter where it is set. If the level is set too low, the signal-to-noise ratio suffers, because the output amplifier introduces hiss in trying to maintain unity gain. If the level is too high, the unit clips, distorting the signal. Sabine recommends starting with a two o'clock position, which proved a good working level for us. A 5-segment LED meter displays levels from -24 dB to clipping.

A hardwire Bypass switch takes each channel out of the signal path. When the device is in Bypass, it has no effect on the program and simply passes the signal, even if the unit loses power. However, if you combine balanced and unbalanced connections through the unit (something Sabine does not recommend), you may notice some coloration or loss of level in the program material.

FILTER CONTROLS

The Filter Control section includes a set of five buttons that access the main features for each channel. The nine Filter Activity LEDs light as each filter is set. The most recently set filter's LED flashes.

After powering up the unit, you can initialize the filter "characteristics" if your preferences differ from the factory defaults. The three initializing controls (Reset, Set Total Number, and Set Fixed) need to be depressed for four seconds before they become active. This is an ample amount of time to safeguard against accidental bumps.

The Set Total Number control allows the user to define the total number of filters to be used. The factory default uses all nine filters, but sometimes you might want to use fewer active filters in order to maintain the original integrity of the source. After you depress the button for four seconds and then release it, the Filter Activity LEDs begin to light sequentially. When the desired number of filters are lit, press the button again to save the setting to memory.

Don't touch that dial! Sabine's FBX-1802 Dual Feedback Exterminator automatically controls feedback using both dynamic and fixed narrow-width filters. The unit offers more user control than earlier members of the FBX family.

The unit automatically resets, clearing all the previous filter-frequency settings.

The Set Fixed control allows the user to select the number of fixed filters versus dynamic filters. A fixed filter, once activated, does not change its frequency center point until the unit is reset. These are the first filters to be engaged, and they eliminate the most prominent feedback frequencies. Although the frequency remains fixed, the amount of attenuation can be increased if the unit senses more feedback at the designated frequency.

A dynamic filter searches for intermittent feedback that crops up during a performance, releasing its previous setting and activating at the current feedback frequency. The 1802 continuously cycles through the dynamic filters, resetting them sequentially. The factory default, which we found suitable, is six fixed filters and three dynamic filters.

Initializing Set Fixed is accomplished using the same procedure as Set Total Number. This is a great improvement over the FBX-900, which has to be powered down before you can reset the number of fixed and dynamic filters, a very inconvenient procedure if your power amps are already on.

In addition, you can change the Set Fixed parameter during the program without clearing all the current filter-frequency settings. On one occasion, we were using the factory default of six fixed and three dynamic filters. During the show, after the six fixed filters were set, one other troublesome frequency kept ringing. We were able to reset the seventh filter as fixed, eliminating the ring with no discernible change in the overall sound.

The red Reset button clears all the filter frequency settings for the selected channel, but does not disrupt the Set Total Number or Set Fixed settings. Once again, the button needs to be pressed for four seconds before the unit resets. When you power down the FBX-1802, it stores all filters and their positions in its nonvolatile memory. The unit returns to its last settings when power is restored. (There is no other way to save user programs.)

The Locked Fixed control permits you to maintain the amount of cut for each fixed filter. Engaging the Locked Fixed button freezes all the fixed filters at their present level. After that, even if the unit senses more feedback at a currently set frequency point, it does not increase the attenuation. If only three fixed filters have been set and Set Fixed is engaged, the remaining fixed filters become inactive; however, Set Fixed does not affect the dynamic filters. This new feature solves one of our major concerns with the FBX 900 by removing the "do-or-die" automation and allowing the user to determine the desired amount of filtering. This is especially critical in music applications, where the FBX unit can mistake sustained synthesizer or organ notes or guitar feedback as system feedback.

▼

Those narrow filters are the heart of the FBX series.

Finally, the Fifth Octave control gives you the flexibility of widening the bandwidth of any or all of the filters. This is helpful in spoken word or "talking-head" applications, where the smoother cut of 0.2 octave often sounds more pleasing. Only filters engaged after pressing the Fifth Octave button will set with the wider bandwidth. Press the button again, and any subsequent filters are set at the standard 0.1 octave.

POSITIVE FEEDBACK

The FBX-1802 is a successful refinement of Sabine's early feedback exterminators. The company has provided twice the capability in the same-sized package and has added critical new features. At times we relied on the 1802's maximum power, while at other times it was simply a safety net. With our busiest season upon us and the crew stretched thin, the FBX-1802 was a godsend. Over the course of a week, we used it on several shows, all on different systems. In every case it performed remarkably well.

One opportunity was a typical corporate production with a Meyer loudspeaker system, podium mics, and lots of wireless lavaliere mics. We inserted one channel of the unit on the podium submaster and the other channel on the lavaliere submaster. With very little time or assistance in setting up, one of us was able to ring out the microphones with the FBX-1802 in just a matter of minutes. Both sets of mics remained stable and provided plenty of gain. The audio quality was much better than trying to ring out the mics with the channel EQ or inserting a ⅓-octave EQ. The narrow bandwidth of the 1802 filters, even at the 0.2-octave setting, caused less coloration than with a conventional equalizer, and the channel EQ was still available to tailor the sound.

During the Hilton's entertainment night, the monitors needed to be mixed from the house console. The room needed to be converted from the general session meeting in about two hours. The FBX-1802s were repatched into the auxiliary monitor sends and once again saved our butts. We were still plugging in microphones five minutes before the doors opened, so we quickly rang out two mixes of vocal mics, and sound check was over. We had no feedback problems for the entire gig, and the band commented during break and after the show on the great-sounding monitors.

FBX-1802 Specifications

Frequency Response	20 Hz-20 kHz
S/N Ratio	>100 dB
THD	<0.02%
Dynamic Range	>100 dB
Max. Attenuation	40 dB
Response Time	approx. 0.4 sec.

Product Summary

PRODUCT:
FBX-1802 Dual Feedback Exterminator
PRICE:
$1,299.95
MANUFACTURER:
Sabine, Inc.
4637 NW 6th St.
Gainesville, FL 32609
tel. (800) 626-7394
or (904) 371-3829
fax (904) 371-7441
CIRCLE #446 ON READER SERVICE CARD

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

UNDER CONTROL

Sabine has made believers out of us. They have resolved most of the issues we had with the older FBX-900, especially the lack of control it offered the sound engineer. Admittedly, the fact that the FBX series is automatic is what makes it so appealing, but having absolutely no say in how the older unit responded was just a bit damaging to our already fragile egos. New features such as being able to lock the fixed filters at their current settings and increasing the bandwidth to 0.2 octave puts the engineer back in charge.

At last, we engineers have been relieved of our unwanted feedback-control burden yet left in command of our system's sonic integrity. The FBX-1802 belongs on your short list of must-have gear for serious sound-reinforcement.

Lori Bolender *has been technical director at the San Francisco Hilton for seven years and recently had her second child.* **Mike Cutter** *has been a sound engineer at the Hilton for six years and nearly had his first nervous breakdown during Lori's maternity leave.*

DigiTech Studio Vocalist

By Rob Shrock

Clone your singers and tune 'em up after they've gone home.

Wouldn't you know it? The lead singer just belted out a great take except for one sharp note in the middle of the chorus. She could do another take, except the last one blew out her voice completely. To top it all off, the backup singers haven't showed up yet, and this track is supposed to be finished tonight. What will you do?

You could join the Foreign Legion and forget all this recording stuff, or you could rescue the vocal tracks with the DigiTech Studio Vocalist. This pitch-shifting device can actually fix out-of-tune notes and generate up to 4-part harmonies from a single vocal input. It recognizes the incoming vocal pitch and creates musically correct harmonies based on standard scales and chords, a user-defined map, or MIDI notes played on the fly.

PROFESSIONAL FEATURES

The Studio Vocalist is based on the technology of the earlier VHM-5 (a.k.a. The Vocalist; see the review in the April 1992 **EM**). Although these products are similar in operation, there are some major differences that make the new Studio Vocalist a truly professional unit. For example, the original VHM-5 was a tabletop model; the Studio Vocalist is a 2U rack-mount device. Internal processing is more powerful in the new model, featuring a 48 kHz sampling rate and 18-bit Delta-Sigma A/D converters. It has a beefy, built-in power supply (no wall wart!), which often means that a company has taken more care in other design areas, too. The Studio Vocalist is sturdy and well-designed to provide easy access to its many features.

DigiTech designed the Studio Vocalist to accommodate both the home and professional studio. On the back panel are two XLR input connectors, one for line level and one for mic level. In addition, there is a ¼-inch, balanced, TRS line-input jack which is switchable between +4 dBu and -10 dBV. (The original Vocalist only provided one XLR and one ¼-inch, unbalanced input.) MIDI In, Out, and Thru connectors also grace the back panel.

The front panel provides an additional XLR mic input that is summed with the back-panel mic input. Mic and line inputs cannot be accessed simultaneously; a button on the front panel toggles between the two. Another button activates the internal +48V phantom-power supply, which means the Studio Vocalist could also be used as a mic preamp. (It's good enough for live use, but I would recommend a high-quality outboard preamp for the studio.)

One limitation of the original VHM-5 was because of its small size; it only had room for a stereo pair of unbalanced, ¼-inch, processed outputs and one ¼-inch, dry output. The Studio Vocalist provides a separate XLR and ¼-inch TRS output jack for each of the four harmony outputs and for the dry, unprocessed input signal. In addition, the XLR and ¼-inch outputs are simultaneously active and affected by the position of the +4/-10 switch. If you do not wish to process each harmony separately or

you have only two channels to spare on your mixer, two of the outputs serve as stereo outputs, and you can program each harmony's level and pan position.

A Send/Return effects loop on the back panel consists of two ¼-inch TS jacks. This is preferable to the more common 1-jack TRS effects loop found on many space-conscious processors and mixers because no special Y cable is necessary. If you have a 3-button footswitch, such as the DigiTech FS300, you can plug it into the back-panel footswitch jack to control Bypass On/Off and one of the following options: Program Up/Down, Song Up/Down, Song Step Up/Down, or Data Up/Down. If you have a simple 1-button footswitch, you can turn Bypass on and off.

An expansion port for future hardware upgrades is included. A company rep told me there are immediate plans for only one upgrade card, but it is an important one. The Studio Vocalist Digital I/O Card ($279.95) has AES/EBU and S/PDIF ports (on XLR and RCA jacks, respectively) and supports sampling rates of 32, 44.1, and 48 kHz. Special software is included that upgrades the Studio Vocalist user interface with new menus for selecting sample rate, synchronization, and input and output formats. The card is slated for release by September 1.

Above the front-panel mic input are the input- and output-level knobs. Four-segment bar-graph meters indicate the input and output levels, but they have a pretty rough resolution of -30, -20, -10, and 0 dB. According to the manual, you are 3 dB below clipping when the 0 segment lights. An Overflow indicator lights when the input signal clips. In the center of the front panel is a large data-entry wheel. Unfortunately, it feels like the flimsiest part of the whole unit, which is surprising considering how often it will get jacked around in ordinary use.

Programming the Studio Vocalist is easy and intuitive thanks to eight "soft buttons" and two arrow buttons under the well-lit, 4-line by 20-character LCD display. Edit, Exit, and Store functions are initiated on the front panel, and there is a dedicated Bypass button next to the input and output knobs.

FADE IN THE BACKGROUNDS

This box gives you instant background-vocal gratification. There are many great programs in the 99 ROM presets. (There are also 99 user memory locations.) Select a Program and sing a note; when you hit a pitch that corresponds in some way with the program (more on this later), a Lock light illuminates, indicating that the processor has identified the incoming pitch and is harmonizing it. Depending on the specific program, virtually all notes will be harmonized, as long as the input signal has a recognizable pitch. If you wish to change the key in which you are singing, just push the corresponding button on the front-panel mini-keyboard. The LCD displays the new tonal center as the harmonies change to match the new key. It sounds like you have four very smart background singers.

Okay, enough instant gratification; it's time to explain what's happening in this box. Each Program uses one of five Harmony Styles: Chromatic, Scalic, Chordal, Vocoder, or Pitch Correct (my favorite). Chromatic harmonies are nonintelligent, parallel harmonies that are always transposed by a fixed interval from the input note (great for parallel, quartal harmony *à la* old Emerson, Lake & Palmer). Chordal and Scalic

The Studio Vocalist offers more professional features than the original VHM-5 Vocalist, including improved audio quality. Aside from its impressive intelligent harmony generation, the unit's pitch-correction features alone make it a big winner.

harmonies are based on common chord types and scale types (see the sidebar, "Harmony Styles"). Pitch Correct retunes the input signal to the user-definable reference pitch in the Studio Vocalist.

The Vocoder is not really a vocoder at all. It generates the harmony note or notes based on incoming MIDI notes, and it has seven different modes: One-Note, One-Note Layered (two harmony voices, slightly detuned to produce a doubling effect), Two-Note, Two-Note Layered, Three-Note, Four-Note, and Four MIDI Channels. (In the latter mode, each vocoder input note is received on different MIDI channels, which is great for MIDI guitar controllers.) In addition, a Transpose function reassigns the notes creating the harmony. The output is based on the timbre of the input signal; only the harmony pitches are determined by note numbers via MIDI.

Programs can be assigned to Steps in a Song to define the chord changes. You can then simply advance through the Steps in the Song with a footswitch as you sing. The Studio Vocalist can hold up to 50 Songs in memory.

A wealth of possibilities presents itself from the very first page of the display. For example, if you call up a Program based on Chordal Harmony, eight of the ten available chord types (in addition to any custom harmonies you may have programmed) are displayed in the LCD window and correspond to the eight soft buttons. You can select a key from the minikeyboard, sing a note, push the "maj" soft button, and (depending on other programmable factors) a fully realized chord will sound. While still singing the note, you can select a new chord root from the minikeyboard, and the harmonies instantly change, glitch-free. (Yes, glitch-free.) Now, press the "dom 7" soft button, and a seventh will magically appear in your harmony. All this, and we haven't even pressed the Edit button yet.

EDITING

When you press the Edit button, eight soft-button options lead you to a full set of editing features. Each soft button selects one of eight editing areas: Gender, Detune, Vibrato, Scoop, Timing, Mix, Name, and Harmony. After that, the Edit button accesses the appropriate parameter pages with the help of the arrow buttons.

The Gender feature simulates the overtones of a male or female voice. It can be programmed to affect any, all, or none of the voices with either Gender setting. I found this feature moderately useful, though somewhat unnatural-sounding.

You can detune each harmony part in 1-cent increments from 0 to 100 cents. A Randomize function applies a different detune amount (within a user-definable range of 1 to 25 cents) to every new harmony note. Pitch Randomize continuously varies the pitch of a note throughout its duration according to one of three settings: high, medium, or low.

The Vibrato function includes several parameters. Depth can be set from 0 to 100 cents for each voice. Rate has a range of 0 to 9.9 Hz in 0.1 Hz increments, and Type offers a choice of sine, square, sawtooth-up, or sawtooth-down waveforms. Delay before vibrato begins is expressed in 20 millisecond increments between 0 and 2.5 seconds, and a Randomize function varies the depth and rate of each consecutive note within an arbitrary range of 1 to 10 that can be set independently for each harmony voice.

Scoop bends each note up to pitch from below (as many singers do stylistically). This function can be programmed two ways: through Randomize or MIDI Velocity. Scoop Occurance determines how often the notes scoop up.

Harmony Styles

Some harmony voicings are based on common chord or scale types. The key center is chosen from the minikeyboard, while the type is selected from a preset list.

Chordal Harmonies	Scalic Harmonies
major	major
major seventh	minor
minor	whole tone
minor seventh	diminished
dominant seventh	blues
minor seventh flat five	dorian
diminished seventh	harmonic minor
augmented seventh	melodic minor
suspended	suspended seventh

It is set between 0 and 100%, with 50% being about every other note. Scoop Amount is expressed in cents from 0 to 900 (nine semitones). Scoop Rate is given in arbitrary units from 1 to 10, and Scoop Threshold is a MIDI Velocity number that activates the Scoop function in Vocoder mode.

The Timing function specifies a delay between each input note and the onset of the harmonies, which slightly staggers the sound of the "background group." The Timing Delay can be set between 0 and 80 milliseconds using 5-millisecond increments. The Timing Randomize feature applies a different delay to each input note.

The Studio Vocalist's Mix function sets the levels and panning of each voice in Stereo mode and sets the wet/dry mix. In Quad mode (which sends all harmony parts and the input signal to their own dedicated outputs), the Mix function sets the output levels of the harmony voices only.

Most of the action is found in the Harmony editor, where you specify the harmony intervals above and below the input note. You can specify one note to remain on the root of the chord regardless of the input pitch and can set Pitch Bend and Portamento rates for each part. The Harmony Motion feature sets break points at which the chords revoice themselves to stay within certain ranges (Normal mode), or they can maintain constant intervals (Fixed mode).

In addition to the Scales and Chords mentioned earlier, Custom Harmonies can be programmed whereby each input note can generate any harmony parts on any pitches (within a range of four octaves up or down).

Bending, which can be set to Stepped or Smooth, is a particularly nice feature. In Stepped mode, each harmony note moves in discrete steps to the next note and does not follow the sliding of the input note. Smooth mode allows the harmony voices to glide between notes as the input note does. Both settings worked well, but Stepped mode sounds better when only one harmony is being created. When you're trying to lock down three or four harmonies that have been generated from an input vocal with a lot of sliding between notes, Stepped mode does a hard quantize on the pitches, which sounds unnatural. If you use Smooth mode in this application, the character of the harmonies should better match the original vocal.

AUDIO PROCESSING

In the Utilities section, there is an assortment of features to help the Studio Vocalist track your vocal input with a minimum of errors. There is an Ess Sensitivity control that acts as a de-esser to the input signal. This helps the processor lock onto the fundamental pitch more quickly by removing excess sibilance, but it does not affect the dry output signal or generated harmonies. This works very well.

Bass Rejection Threshold rolls off some of the input's bottom end but only to the processor, not in the outputs. This helps decrease tracking errors that could be caused by wind noise, microphone handling, background noise, etc.

The Harmony Gating Threshold specifies the level of input that triggers the harmonies. This is designed mainly to prevent band noise from the stage from falsely triggering the unit. I found this threshold to be finicky and sometimes difficult to set for consistent operation, but it was somewhat helpful. I've found several other DigiTech processors to be challenging when it comes to setting proper gain structure, but the Studio Vocalist is quite forgiving in this area.

An Anti-Feedback Control attenuates high frequencies into the processor, though I did not have the opportunity to test this feature. The Tuning reference produces a nominal A at 440 Hz and can be adjusted up or down up to 100 cents.

PITCH CORRECTION

One of the most useful and well-implemented features of the Studio Vocalist is its Pitch Correct function. I am currently working on an album that involves a singer who lives in another city and isn't available for endless vocal takes. Like most projects, there have been a few lines the vocalist delivered with great feeling but with less-than-accurate pitch. In particular, one song had an overall pitch problem.

I tried out the Pitch Correct capabilities on an ADAT vocal track, and I was amazed! Without a glitch, the Studio Vocalist tightened up the tonal center of the track, yet the sound was not colored by the process. The slightest bit of top end disappeared, but a very small boost at 12 kHz restored the signal to its original frequency content with no other side effects except a tiny—and I do mean tiny—increase in noise. I would have no problem running a whole performance through the unit. I would also use it for occasional lines; the processed signal is so clean that any difference from the original signal is negligible. In addition, I was surprised at how well the unit can take a signal with a heavy vibrato and still lock it into pitch.

In addition to the lead vocal track, I had two background tracks with similar pitch problems on the ADAT. I sent all three tracks through the Studio Vocalist to see what would happen. Although the unit is designed for single-note input, it handled all three parts simultaneously without a hiccup. I wouldn't recommend using it this way, but it's nice to know the processor has this much power.

In the course of this experiment, I mixed the original and processed lead vocal tracks, which produced the most naturally gorgeous doubling effect I've heard. Most pitch shifters detune an existing track to create the effect of a doubled part. The Studio Vocalist takes the naturally occurring pitch deviations and brings them more in tune. The result still gives you doubled effect, but with a more focused, in-tune sound than any processor that uses the detuning method (including some high-end professional processors).

I checked this on some very in-tune performances by other singers, and the result was still the pleasing effect of a double-tracked voice, similar to the sound of a singer who doubles himself

Product Summary

PRODUCT:
Studio Vocalist
PRICE:
$1,049.95
MANUFACTURER:
DigiTech
8760 S. Sandy Parkway
Sandy, UT 84070
tel. (801) 566-8800
fax (801) 566-7005
CIRCLE #447 ON READER SERVICE CARD

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

really well already. If someone out there sings so well that he or she can make this unit sound bad in this application, call me; I'm ready to produce your album.

MIDI DOES HARMONY

The Utilities section provides access to the MIDI parameters. Program Changes let you select Programs that match the chord structure of the song. As mentioned earlier, the Vocoder modes use the chords you play on a MIDI controller to determine the absolute pitches of the harmony parts. Most of the Vocoder modes expect to see MIDI note messages on one channel, while the Four-Channel Vocoder uses different MIDI channels for each of the four harmony input notes.

One particularly interesting and useful parameter is Harmony Hold. Sending the appropriate message freezes the harmonies on their current notes, allowing the input vocal to riff. I tried this with a prerecorded vocal that performed a trill after establishing a note. The Studio Vocalist locked down the harmonies before attempting to reharmonize the input during the trill. I occasionally got some strange artifacts when using this feature with prerecorded vocals and found it a bit less reliable than the other Studio Vocalist features. As a result, I suspect it would be more difficult to get consistent results in live performance than in the studio.

You can modulate volume, detuning, vibrato depth, vibrato rate, and Harmony Hold using any Control Change message. You also can store and retrieve Programs and/or Songs via SysEx. In addition, the buttons on the front panel send different values for Control Change 16. Recording basic moves into a sequencer is fast and painless, especially if they include no more than changing a chord root or type. DigiTech has really kept the end user in mind with this unit.

CONCLUSION

There is plenty to like about the Studio Vocalist. Although "intelligent harmonies" still have a somewhat strange and unnatural sound over extended periods of time, this unit does them better than any other I've heard. The output is clean, virtually glitch-free, and uncolored. It can add extra harmonies to the vocal tracks after the singers have gone home. In a live setting, your vocal sound can be expanded and still make musical sense, if you don't mind spending some time programming. The editing capabilities are enormous, yet the user-interface is simple and intuitive. The manual is well-written and even includes a guide to basic harmony.

The biggest downer is the price: at just over $1,000, it's not cheap. On the other hand, considering that the original VHM-5 retailed for $849, the Studio Vocalist's professional I/O design, additional features, and cleaner audio are well worth the extra bucks. The Pitch Correct function is so useful that it almost justifies the price tag all by itself. What's more, if you are in need of a doubler, the Studio Vocalist stands in a class of its own.

Composer and producer **Rob Shrock** *is the keyboardist/arranger for Dionne Warwick and Burt Bacharach.*

CLASSIFIEDS

ELECTRONIC MUSICIAN CLASSIFIED ADS are the easiest and most economical means to reach 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 CONSULTING

ACOUSTIC FOAM

Don't let a bad room ruin your sound. Our industry-leading foams can make any space sound world-class at prices any studio can afford. FREE SHIPPING! Is your room leaking sound? Our SheetBlok Barrier is what you need. Moving soon & don't want to leave your foam behind? Get our Vel-X panels. We also have absorbent fiberglass, adhesives, bass traps, etc. Clients: AT&T, NASA, Sony, Warner Bros., 20th Century Fox, Ford, Hitachi, Dennis DeYoung, Toshiba, Universal Studios, Maury Povich Show, NPR & many more. We CAN improve your sound & save you big money!

NEW **Acoustics 101** The world's cheapest, easiest to understand booklet on how to turn your space into a recording studio. Learn how to build your walls, float your floor, seal off your windows, box in your air conditioner & more! The low price includes postage & you get a $10 coupon for when you buy foam! Send check or money order today!

Just $9.95

USAFoam 11571 E. 126th Fishers IN 46038 · Info (317) 251-2992
Orders Only (800) 95-WEDGE · Fax Line (317) 257-2USA

EMPLOYMENT OFFERED

Recording Engineer
Radio Announcer

ON THE JOB TRAINING
IN MAJOR LOCAL RECORDING STUDIOS/RADIO STATIONS
Part time, nights, weekends. No experience required. Free brochure & recording tells how.
1-800-295-4433

COMPOSER WANTED
Top LA music production house looking for a musical renaissance man/woman who can compose in any genre and make it sound powerful. We need a true synth master to help launch a new division, scoring TV and radio commercials, film, CD-ROM, you name it. We offer an excellent environment, quality people, full staff, complete benefits. This is a hot career opportunity for the right person. Send tape and resume to: 4700 Los Feliz Blvd. #18, LA, CA 90027. No overnight packages please.

Assistant Engineer
Hollywood recording studio seeks Assistant Engineer. Neve & GML experience helpful. Tel. (213) 851-5228.

PROFESSIONAL MUSICIANS REFERRAL
Musicians/Bands: Connect with the RIGHT group or player. Call PMR today; save time & money. PMR—America's original national referral!
(612) 825-6848

EQUIPMENT FOR SALE

OMNIRAX
STUDIO FURNITURE for the discriminating musician

800-332-3393
P.O.BOX 1792 SAUSALITO, CA 94966

MOBILE MIDI: Carriers for MIDI percussion (SPD.11) or keyboard controllers. Complete freedom of mobility for live performance! No cables. Battery-powered, wireless. Options: MIDI cymbals, "Triggershoes." "Get Up and Go Walk-about!" WALK-ABOUT INC., PO Box 66058, LA, CA 90066. **(800) 430-WALK**. E-mail: GO4WALK@ AOL

Sound thinkin
The Best Prices, The Best Service
you'll ever find on...
digidesign
ALESIS
and related products
503-386-2682

PLUG IN!
Innovations for the Electronic Musician and Percussionist. We carry Akai Samplers, TB-303 BassStation MIDI Analog keyboard, Mackie mixers, Drumatic Hand Drum Triggering System, and experimental percussion instruments. We also buy and sell old Roland drum machines and Analog keyboards. Call Mandala Plugged (800) 858-2822 for info.

AVR
AUDIO VIDEO RESEARCH SYSTEMS, INC.

PROFESSIONAL AUDIO EQUIPMENT
NEW and USED
• *PROFESSIONAL ADVICE*
• *THE RIGHT PRICE*
BOSTON
(617) 924-0660
CONNECTICUT
(203) 289-9475

SINGERS! REMOVE VOCALS
Unlimited Backgrounds™
From Standard Tapes, Records, & CDs with the **Thompson Vocal Eliminator™**
Call for Free Demo Tape.
Phone (404)482-4189-Ext 16
LT Sound, Dept EM-1
7988 LT Parkway
Lithonia, GA 30058
24 Hour Demo Info Line (404)482-2485 - Ext 16
Best Vocal Eliminator™, Key Changer, Voice Enhancer!

Nadine's
MUSIC & PRO AUDIO
The Right Advice,
The Right Price,
Right Here,
Right Now!
800•918•4800
ALL MAJOR BRANDS.
CALL US!
(213)•464•7550
6251 Santa Monica Blvd • Hollywood,CA • 90038

CLASSIFIEDS

NEW, USED, DEMO EQUIPMENT
BEST SELECTION OF DIGITAL/ANALOG RECORDERS, CONSOLES, DAWS, OUTBOARD GEAR

Otari Status 18R, Trident, Soundcraft DC2000, Mackie 8•Bus, Neve 5432, Otari Radar, MTR90II 24-trk, 34C 40-ch w/p. bay, Concept 1, Neve 8108, Pro Tools III, Avid AudioVision, Sonic Solutions, ADATs, TASCAM DA-88, Apogee AD1000, Lexicon PCM 80, t.c. Electronic M5000, Roland RSS-10, API, GML, ADL, Summit, Focusrite, Demeter, Drawmer, TL Audio, Neumann Mics, AKG C12VR, Microtech UM92SZ, B&K Mics, KM 184.

Studio and System Design, Financing, Factory Service, Installation. Experienced and Knowledgable Sales Staff.

EAR PROFESSIONAL AUDIO
(602) 267-0600

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

INFORMATION: 1(800) 427-5698
Outside U.S. 314 346-8549, Fax 314 348-2769
ACI • Rt 3, Box 4374-A • Osage Beach, MO 65065

Don't Get Beat
When you need equipment call

8TH STREET MUSIC

(800) 878-8882

Philadelphia's Largest Musical Instrument Dealer!!!

8th Street Music, 1023 Arch St.
Philadelphia, PA 19107

Hard to find **MEMORY** for your **AKAI SAMPLERS**. Memory, SCSI Interface for most Samplers. Hard Drives, Pro Audio, Cables.

(800) 979-9066

Mountain International
501 4th St. SE, Bandon-by-the-Sea, OR 97411. Tel. (503) 347-4700; fax (503) 347-4163.

Music Software & Hardware

All Brands/All MIDI Products. Specialists in Keyboards, Samplers, and Digital Audio.

(800) 977-9789
YRS MIDI Systems,
Orlando, FL

Wanted For Cash

Old synthesizers, drum machines, sequencers, etc. Any condition, dead or alive!! Especially Moog, ARP, EMS, EML, Oberheim, Hohner, Vox, PPG, Roland. Will airmail you cash. Absolute best prices paid. Echo Park Music & Sound, PO Box 486, Park Ridge, NJ 07656. (201) 930-9660/fax: (201) 930-9678.

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.** 66 years in business. **Worldwide delivery**. Visit our new **22,500 sq. ft.** location. **Call**, write, or fax us today for price quotes and details. Also, find us on the Internet at **sales@caruso.net**. Start saving money today. Call **Caruso Music**, 94 State St., New London, CT 06320 USA. **(203) 442-9600**/fax: (203) 442-0463.

Roland DM-800 Digital Audio Workstation. 8-trk., no computer needed. 6 months old, perfect cond. $4,000.00. (702) 631-0797.

CLASSIFIEDS

EQUIPMENT FOR SALE

Wanted: Roland RSP-550 signal processor, Roland A-110 MIDI monitor, Miditec UBM4 key interface, Lake Butler MSI-8 relay, Simmons MTM. Yamaha DMP9-16. **For Sale:** Peavey PLM 8128 8-chan. MIDI-controlled mixer + PLM 8128E expander. (415) 574-1987 lv. msg.

CASIO CLOSEOUT!
Digital MIDI Horn
$99.95
VZ-10M Synth Module
Priced with
2 FREE
ROM Cards at
$299.95
(414) 784-9001

The Stick®—used from Stick Enterprises, Inc. Optimum setup and action as with our new models. Call Emmett Chapman for newsletter, literature, and brochure. (818) 884-2001. (No affiliation with Traktor Topaz, also known as Arthur Cronos Action Marketing.)

WORLD'S SMALLEST EQUIP. DLR. GET ON THE MAILING LIST NOW! R-8, $375; Quadra-Verb, $275; D-50, $600; S900, $650; M1R, $700; TSR-8, $1,500. Mics, Mixers & Processors aplenty. Call for today's availabilities. TELESIS (714) 998-3001.

Looking for new/used recording, keyboard or outboard equipment? We've got **tons** of super clean **used**, as well as **new** products from **TASCAM, Alesis, Kurzweil, Roland, Mackie, Genelec, Lexicon, Fostex, Yamaha, Korg**, and hundreds more. **Discount pricing** and **Worldwide Delivery!** 66 Years in business. Trade-ins welcome. Visit our new **22,500 sq. ft.** location. Call, write, or fax us today for price quotes and details. Also find us on the Internet at **sales@caruso.net**. Start saving money today! Call **Caruso Music**, 94 State St., New London, CT 06320 USA. **(203) 442-9600**/fax: (203) 442-0463.

Chapman Stick, Warr Guitar. Buy, sell & trade. Free newsletter subscription, lessons, cassette reviews. Strings, supplies. Credit cards OK. Traktor Topaz. 24-hour recorded info: (415) 435-7504.

INSTRUCTION

BACHELOR OF SCIENCE IN MUSIC TECHNOLOGY
Study sound synthesis, MIDI, digital recording, and electroacoustic composition in a comprehensive School of Music. For more information contact: Prof. Jeffrey Stolet, School of Music, 1225 University of Oregon, Eugene, OR 97403-1225. (503) 346-4094. E-mail: stolet@oregon.uoregon.edu

"Musicware will teach you the skills to make music for a lifetime."–*PC Magazine*

PC Magazine gave its highest rating to Musicware Piano. ★★★★ Four Stars–Excellent! For information on America's best-selling, hands-on music education software, call **1-800-99-PIANO.**

Musicware.

Music Engineering Technology, Bachelor of Science Degree. The only program in the country where you can learn MIDI from A to Z, synthesizer and sampler hardware, digital audio, and software design. Fully equipped individual MIDI workstations. Careers for music, electronics, and computer industries. Accredited. **Cogswell College**, 1175 Bordeaux Dr., Sunnyvale, CA 94089-1299. (408) 541-0100.

ATTENTION: Keyboardists! Beg-Pros. **FREE** Report reveals how to sound like your favorite keyboardists on recordings. **(800) 548-5601**, 24 Hrs. **FREE** Recorded Message.

WANTED: TRAINEE FOR RECORDING ENGINEER
On the job training at major studio **IN YOUR AREA.**
Keep present job, no experience required.
Train around your own schedule.
THE RECORDING EXPERIENCE
Call: 1-800-795-5750

Frustrated
with owner's manuals?

Get a free copy of our catalog, which features tips and techniques books for equipment by Roland, Alesis, Yamaha, Casio, Korg, Ensoniq, Kawai, Kurzweil and Oberheim.
All titles reviewed and guaranteed!

MIX
BOOKSHELF
6400 Hollis St. #10
Emeryville, CA 94608
(800) 233-9604 • (510) 653-3307
e-mail: mixbooks@mnusa.com

Be a recording engineer. Train at home for a high-paying, exciting career, or build your own studio. **Free** information. **Audio Institute of America**, 2258-A Union St., Suite F, San Francisco, CA 94123.

LEARN THE ART OF RECORDING
Learn the practical skills needed to start a career as a engineer, producer or studio musician. •300 hours •Six studios/latest equipment •Small classes •Job placement assistance •Financial aid •On-campus housing
Call 800-848-9900 or 614-663-2544
The Recording Workshop
455-L Massieville Rd, Chillicothe, Oh 45601

PARTS & ACCESSORIES

BIG DISC
DATA STORAGE SYSTEMS
RACKMOUNT PORTABLE
REMOVABLE CD-ROM
PROFESSIONAL STORAGE SOLUTIONS FOR DAW'S AND SAMPLERS
DIGIDESIGN ENSONIQ MAC
AKAI ROLAND IBM
YAMAHA EMU ATARI
PLAY HARD.....SPIN BIG.
(305) 749-0555
MON-FRI 10AM-7PM EST. VISA / MC

KEYBOARDS & PROAUDIO
SCSI DRIVE SALES/REPAIR
CD RECORDER
4X • ALL BOOKS
$3995
Generalmusic
S2 & S3 Turbo
MUSIC PROCESSOR
SX2 & SX3
MULTIMEDIA WORKSTATION
INFO & FREE CATALOG
619-789-6558
ORDERS 800-753-6558
FAX 619-789-4170
1125 ELEVENTH ST
RAMONA, CA 92065

SOUND LOGIC
EXPANDERS

Ensoniq Seq. Mem	157
Akai MPC60 RAM	174
EPS/EP16 SCSI	147
Akai S950 RAM	174
S1000 8MB	594
EPS 4X RAM	247
W30 SCSI I/F	174
Roland M256E	74
Korg MCR-03	74
Yam MCD-64	74

TYPHOONOS TX16W *164
SAMPLINGCDS
$47.95
FREE SHIPPING

PUBLICATIONS & BOOKS

SONGWRITER PRODUCTS, IDEAS, NECESSITIES! Contracts, copyrights, books, critiques, record company/publishing/radio directories, bio kits, photos, cassettes, labels, MIDI, studio supplies, duplication. "Free catalog!" Songwriter Products, 345 Sprucewood Rd. #5, Lake Mary, FL 32746-5917. VISA/MC, Orders (407) 321-3702.

THE GUIDE TO MIDI ORCHESTRATION
FILM COMPOSER Paul Gilreath demonstrates through detailed examples, charts, background, recommended setups, and MORE. How to achieve extremely REALISTIC orchestration from your MIDI set up. **Only $14.95.**
(800) 469-9575

Books, Tapes, Videos
Get our free catalog featuring resources on
- **MIDI** • **Instrument-specific guides**
- **Drum-machine patterns**
- **Synthesis & sampling**
- **Recording** • **Composition**
- **Music business**
- **A&R lists and more**

All titles reviewed and guaranteed!
MIX
BOOKSHELF
6400 Hollis St. #10
Emeryville, CA 94608
(800) 233-9604 • (510) 653-3307
e-mail: mixbooks@mnusa.com

CLASSIFIEDS

RECORDING SERVICES AND REPAIRS

FREE CASSETTE DUPLICATION
Real-Time Ships/3 days. Lowest Prices Guaranteed! Order 90 C-30s and Get 10 **Free** with boxes for $116! We will beat any advertised price! Accurate Audio Labs, Inc. (800) 801-7664.

MAIN MASTERING
•MASTERING
•DAT TO DAT
•CD REFERENCES
•50+ CD PACKAGES
(213) 436-MAIN

****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.

DIGITAL FORCE™
212 333-5953
MASTERING
REPLICATION
PERSONAL EXPERT SERVICE
TOTAL CD, CD-ROM & CASSETTE PRODUCTION
330 WEST 58th ST NY, NY 10019
The Power of Excellence SM
GRAPHIC DESIGN
PRINTING
COMPLETE PACKAGING
POSTERS

CDs IN 3 WEEKS!
500 CDs and 500 Cassettes only $2,590 with B&W inserts
• FREE Graphic Design • Includes Insert Printing
• Major Label Quality • No-Fine-Print Guarantee
Call today for your FREE, 1995 full color catalog:
1-800-468-9353
24 HOURS TOLL FREE
609-663-9030 FAX 609-661-3458
DISC MAKERS

Your music on Compact Disc
Any quantity from 1 to a zillion
Lowest prices, Best service
Big Dreams Studio, Ltd.
708-945-6160

RECORDS, TAPES & CDS

1-800-TAPE WORLD or 1-800-245-6000
We'll beat any price! 5.95 SHIPPING • FREE CAT.

SONY		MAXELL		TDK		FUJI	
DAT-120	7.49	DAT-120	7.99	DAT-120	6.99	SVHS-T120	6.69
DAT-120 pro	9.99	XL11-90	1.69	SA-90	1.59	VHS-T120	10.99
T-120V	2.09	XL11S-90	2.09	SAX-90	1.99	DR-I-90	.69
L-750HG	3.99	T120HGX	2.79	T-120EHG	2.49	DENON	
JVC T120	1.79	ST-120	6.99	HI-8-120	5.99	DAT-120	6.49
JVC ST120	5.99	ST-160	8.99	CDR-74	9.99	DAT-60	4.99

TAPE WORLD 220 SPRING ST. BUTLER, PA 16003 FAX 412-283-8298
OVER 500 DIFFERENT. SAME DAY SHIPPING. M-F 8-5

20th Anniversary **THE WAREHOUSE** Since 1975
Studio Sales & Services
Cassette, CD, CDR duplication
Rt@R · DATs · ADATs · HI8 · VHS · CDs
PC/Windows audio workstations
Warehouse prices
800-483-TAPE · fax: 904-398-9683
Visa · MasterCard · Discover
2071-20em Emerson St., Jacksonville, FL 32207 · 904-399-0424

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

THE MACHINE ROOM
"Let us enhance your mix with digital mastering"
(for as little as $10 a track)
Don't wait 3 weeks for your CD's, call us for the fastest service!
1000 CD's for $1250
500 CD's - $875 1 CD - $30
packages include: Glass Master, 2 color on CD, JewelBox, Test Pressing, Insertion of your graphics, Packaging & Shrinkwrap, add $400 for 4 panel insert, 4 color on front & traycard.
NEW 5 CD's - $100
1000 Promo CD's for $999
24HR 713-781-1963

SAVE 50% on your next duplication order!
WHETHER YOU HAVE SOMEONE PRODUCE YOUR TAPES & CD'S FOR YOU OR YOU WANT TO DO IT YOURSELF, KABA IS THE NAME TO KNOW.
FOR A LIMITED TIME, KABA WILL APPLY 50% OF YOUR NEXT CASSETTE DUPLICATION ORDER TOWARDS THE PURCHASE OF YOUR OWN *KABA 4-TRACK REALTIME DUPLICATION SYSTEM.**
*SOME RESTRICTIONS APPLY, CALL FOR DETAILS.
THE KABA SYSTEM OFFERS YOU:
★ 20-20 kHz FREQUENCY RESPONSE
★ HIGHEST QUALITY REALTIME & 2X OPTION
SERVING MUSICIANS WORLDWIDE FOR OVER 20 YEARS
KABA AUDIO PRODUCTIONS
KABA RESEARCH & DEVELOPMENT
24 COMMERCIAL BLVD., NOVATO, CA 94949
800/231-TAPE • 415/883-5041 • FAX 415/883-5222

All Manufacturing In Our Plant...
EUROPADISK, LTD.
Bulk CD's as low as 75¢
500 - Promo CD's - $995
(1,000 CD's - $1,550)
Delivery in 15 Business Days!
Promotional Package includes: CD-R Reference, One Color CD Label with Layout, Typesetting & Film (to 63 Min.)
PACKAGE SPECIALS - "With This Ad Only"
500 - Cassettes - $715
(1,000 Cassettes - $1,025)
Retail-Ready: FULL COLOR J-Card, Test Cassette, Graphics Layout, Typesetting & Film, Cassette Label, Norelco Box & Shrink-Wrap (to 50 Min.)
500 - CD's, Full Color Graphics - $1,995 (1,000 CD's - $2,265)
Retail-Ready: FULL COLOR Front & Tray Card, free CD-R Reference, 2-Color CD Label, Graphics Layout, Film, Jewel Case & Shrink-Wrap (to 63 Min.)
500 - 12" Vinyl - $1,105
(1,000 12" Singles - $1,510)
Retail-Ready: 12" Single Package Direct Metal Mastering, 5 Test Pressings, Label Layout & Printing, Plastic Sleeve, Black or White Die-Cut Jackets & Shrink Wrap (to 14 Min. per side)
Call For Our Complete Catalog (Incl. Bulk CD pricing)
EUROPADISK, LTD.
75 Varick Street, New York, NY 10013
☎ (212) 226-4401 FAX (212) 966-0456
(800) 455-8555

CRYSTAL CLEAR SOUND TAPEMASTERS
Your music on CD $50
Custom One-off CD-Rs
•••
1000 Compact Discs $1750
1000 CDs/500 Cassettes $2150
From DAT master, includes assembly with customer-supplied inserts
1-800-880-0073 MC/VISA ACCEPTED

CLASSIFIEDS

RECORDS, TAPES & CDS

3M AGFA AMPEX SONY
Audio, video & digital tapes
Sony Professional Pro Audio,
Neumann, Sennheiser, B&K,
AKG, Fostex, Shure
& Crown Products.
R & M PRO AUDIO
691 10th Ave., SF, CA 94118
(415) 386-8400/fax: 386-6036

OUTRAGEOUS PRICES !
C-10 $.25 C-20 $.31 C-30 $.35
C-45 $.43 C-62 $.52 C-92 $.74
BASF Chrome+ or Aurex Cobalt (like TDK SA), 5 screw clear Olamon shell, 100 piece box, bulk. Also Custom lengths, Boxes, Onshell print, etc.
SoundSpace Inc. 1-800-767-7353

MASTERING • MANUFACTURING • PRINTING
DIGI·ROM™
FOR COMPACT DISCS • CD-ROM
REAL TIME & HIGH SPEED AUDIO CASSETTES
COMPLETE PACKAGES • COMPETITIVE PRICING
GRAPHIC DESIGN STUDIO • FAST TURN-AROUND
PERSONALIZED EXPERT SERVICE
1-800-815-3444
(212) 730-2111
130 West 42nd Street • New York, NY 10036

603-352-3403
CD ONE-OFFS
CALL FOR YOUR FREE CD KIT
ONLY $29
Up to 63 Minutes
BLUE PLANET STUDIO

$1.45 a CD!
WAVE
Digitally Duplicated Cassettes
Graphics - Printing
1-800-363-8273
World Audio Video Enterprises

LENGTH	PRICE (ea.)
C-10	$0.30
C-20	$0.36
C-30	$0.42
C-45	$0.48
C-60	$0.58
C-90	$0.78

TDK PRO SA BULK
CUSTOM LOADED
HIGH BIAS COBALT
CASSETTES
(UNLABELLED AND UNBOXED; $25 MINIMUM ORDER)
NOW! Recording Systems, Inc.
32 WEST 39th STREET, NEW YORK, NEW YORK 10018
TELEPHONE: 212-768-7800 ◆ 800-859-3579

Good Vibrations—RJR Digital
500 Bulk CD—$890
500 Finished CD—$1600
1000 Finished CD—$2040
—Call For Details—
Fax On Demand & Customer Service:
1-800-828-6537
619-267-0307 • FAX 619-267-1339

CD Replication, small quantities available, CD premastering, CD-R, direct-from-digital cassette duplication, art support, package pricing on cassette and CD combinations.
(800) 988-3710
LMR Productions, Inc.

TDK PRO SA
Custom Loaded in
Clear Michelex Shells
C10 .29 C20 .35
C30 .41 C45 .47
C60 .58 C90 .78
Visa, MC, Discover
International Audio
708-298-8555

HUMMINGBIRD RECORDING—COMPACT DISC & TAPE MANUFACTURING. 1,000 full-color CD package, $2,147; 500 Color CD & Tape Combo, $2,597. Digital tape dubbing at slow speed. Many other packages. Call for free catalog. (800) 933-9722.

CD STORAGE+
FREE color literature! Solid-wood cabinets for CDs, Cassettes, Videos, LPs & LaserDiscs
(800) 432-8005

TOLL FREE 1-800-835-1362
Incredible CD and Cassette packages
Complete Full Color CD Packages:
500 CD's $1199.
1000 CD's $1649.
Complete Full Color Cassettes Packages:
500 Cassettes $814.
1000 Cassettes $909..
CD Package includes: color 2 page folder, color tray card, 2 color on CD printing, pre master & glass master, jewel box, shrinkwrap, insertion of graphics.
Does not include film or typesetting
HEALEYdisc

CD'S FAST
*300 CD Special: $849
*500 CD Special: $999
*with client provided CD-R label film & inserts.
Digital Mastering, CD-Rom & Graphic Design Available
new
Silicon Graphics Digital Printing
Full color inserts from $275
10 Free Posters w/complete package
Call (800) DIGIDOC
DIGIDOC PRODUCTIONS

PCSW pacific coast sound works
1,000 CD'S FROM:
$1,500!
► SONIC SOLUTIONS NoNOISE™ ► GRAPHIC DESIGN ► PRINTING ► MASTERING ► PACKAGING ► CD-R ► SONY 1630
1 (800) 423-2834
TEL: (213) 655-4771 FAX: (213) 655-8893
8455 Beverly Blvd., Ste. 500 L.A., CA 90048

YOUR AD COULD BE HERE
Call (800) 544-5530

CLASSIFIEDS

SOFTWARE & PATCHES

GENERAL MIDI FILES
Romeo Music MIDI Editions™
630 Potter Rd., Suite 100
Framingham, MA 01701
World's largest library of royalty-free music on disk; Pop, Jazz, Folk, Classical.
Free catalog! (508) 877-8778.

IBM Mac Atari ST C64/C128
Hundreds Of Music/MIDI Software Disks From $3.00 Per Disk! Call Or Write Today For Your Free Catalog Disk! Please Specify Computer Type.
Music Software Exchange
Post Office Box 533334
Orlando, FL 32853-3334
Telephone 407/856-1244

Gravis UltraSound Patches
Pro GUS, ACE, MAX sounds on CD-ROM. Drums, analog synths, rave, techno, loops, real instruments, more. 100s on each CD, $99.95. U.S. (800) 267-HOWL, (613) 599-7927; fax: (613) 599-7926.
Howling Dog Systems
Kanata NPO Box 72071, Kanata, ON, Canada K2K 2E1

MUSIC SOFTWARE Discounts
Mac Dos Windows Amiga C/64 IIe GS Atari
ELECTRONIC MUSICIANS • EDUCATORS
SOFTWARE--SEQUENCING • NOTATION • TRAINING
Any MIDI INTERFACES • Keyboards • Modules
SOUND MANAGEMENT 800-548-4907
P. O. BOX 211 • Lexington, MA 02173 • FAX: 617-860-7325 • Open Weekends! • Price Quotes
Reference $5. 240 pp MIDI BUYER'S GUIDE

KID NEPRO
KID NEPRO IS THE PATCH KING !
SINCE '84 MORE QUALITY SOUNDS, FOR MORE INSTRUMENTS THAN ANYONE IN THE ENTIRE UNIVERSE!

KORG•
01W SERIES 03RW
ALL WAVESTATIONS
T-SERIES M1 M1R
M3R DS8 707
EX/DW8000
EX-POLY 800
POLY 800-2
ROLAND•
JV80 JV880
JD800
JD990 JV1080 U20
MOST D SERIES
U220 MKS70
MKS80 MKS50
JX8P JX3P JX10
JUNO-1 JUNO-2
JUNO-106
JUNO-60
JUPITER-8 JUPITER-8
YAMAHA•
MOST SY, TG, DX/TX
V50 AND FB01
CASIO•
MOST CZ & VZ SERIES
FZ1 FZ10M FZ20M
OBERHEIM•
MATRIX-6 6R
1000 XPANDER
OB8 DPX1
SEQUENTIAL•
PROPHET-5-VS-
2000-2002
KAWAI•
K1 K1R K1M
K4 K4R
ENSONIQ•
MIRAGE EPS
EPS 16+ ASR10
ESQ1 SD1
VFX-SD
EMU•
EMAX EMAX-2
MOST PROTEUS
SERIES SP1200
AKAI•
S900 S950 S1000 S1100
S2800 S3000 S3200
S01 MPC 60 MPC3000
KURZWEIL•
K2000 K2000S

NEW 70 Disk Library for AKAI MPC 60/EMU SP1200
SOUNDS AVAILABLE IN MOST FORMATS
CALL FOR MORE INFORMATION.
SEND $2 FOR CATALOG AND SOUND LISTINGS.
DEALER INQUIRES INVITED
KID NEPRO PRODUCTIONS • PO BOX 360101 (DEPT E) BROOKLYN, NY 11236 • 718-642-7882
Fax-718-642-8385 • E-MAIL Kid Nepro@AOL.com
AMEX • DISCOVER • VISA • MC
The PATCH KING has it all!

SOUND CHASER Music Software
Everything you need to make your computer musical
- MIDI SOFTWARE
- HARD DISK RECORDING
- NOTATION PROGRAMS
- INTERFACES/ SOUNDCARDS

and much more...
Questions Welcome
Order your **FREE** Music Software Catalog and talk to an **EXPERT**
800-549-4371
FAX (707)826-2994

Join the Best!!!
- Over 7500 Members!
- 12 Monthly Newszines
- New BBS Support!
- New Pro Sequences!
- Expanded MIDI Support
- New MUG™ I.D. Cards

Celebrating our 10 Year Anniversary!
INTERNATIONAL MIDI USERS GROUP
MUG™ supports these instruments and much more:
*EPS series *ASR series *TS Series *Mirage *ESQ/SQ series *S900 *CZ *FZ *VZ *Emu II *Emax *K *DW8000 *M1 *Matrix series *Prophet 2000 *MC500 *D50 *S50 *MT32 *DX7 *IBM *Macintosh *Atari ST *Commodore *Amiga & much, much more!!!
☐ 1 Year Membership $20 (Foreign $30)
☐ Lifetime Membership $75 (Foreign $85)
24 Hour MUG™ Line: (212) 465-3430
Send check or money order to:
International MIDI Users Group™
PO Box 615EM
Yonkers, NY 10703-0615

CLASSICAL SEQUENCER
IDAL is unlike other sequencers. It has the tools you need to step-time CLASSICAL music right. Rubato, repeats, accents, fermati acc, rit, cres, dim, and more.
ELDERSOFT
Box 2435, Idyllwild, CA 92549

MIDI SEQUENCES
TRAN TRACKS —quality service, great prices & a large selection of popular songs. All computer formats and most dedicated sequencers supported.
ORDERS: 1-800-473-0797
Fax: (201) 383-0797
Technical Support: (201) 383-6691
Major credit cards
TRAN TRACKS MIDI SEQUENCES
350 5th Avenue #3304 NY NY 10118

L.B. Music Sequences
We sell sequences for Ensoniq, Korg, Roland, Yamaha, IBM/SMF. General MIDI. Call for a free demo kit.
(800) 3-LBMUSIC
(352-6874)

Music Hardware & Software For Your PC
-Interfaces -Cables -Sound Cards -Software -Accessories
800 524 6073
Visa, Mastercard, American Express and Discover Accepted

SEQUENCES
Finest Sequences & Documentation Available.
Most Computer & Dedicated Sequencer Formats
DAT & Cassette Format
Top 10 Country Hits & Top 10 Pop Hits Always in Stock
Orders: 1-803-293-3767
Technical Support: 1-803-293-4598
Ask About Our Membership Plan
TrackBusters, Inc.
P.O. Box 20279
Myrtle Beach, SC 29575

MIDI JAZZ COMBO. New disk, new songs, check out our latest GM-format jazz sequences. Now only $19.95. **Musicraft Studio**, PO Box 1272, Laurel, MD 20725. Call (301) 604-6297, 24 hrs. Visa/MC.

Give Your Act A Good Kick In The Gas...!
Now In Our 4th Year
Over 5,000 sequences Available
Pop, Country, Classic Rock, Jazz, Big Bands, Standards, Top 40, Gospel, & Italian
Free Catalog
Now The Largest Sequence Company In The World...
Supporting many formats including General Midi.
Midi Hits
3 Shratton Ave. ★ San Carlos, CA 94070
1-(800) 593-1228
Fax (415) 637-9776
Int'l (415) 637-9637

CLASSIFIEDS

SOFTWARE & PATCHES

Turn your K2000 into a **TB303!** Includes sequences, sounds, and instructions on how to make your own loops, **$30.** Also, **500+ K2000 synth patches, only $25!** Send $3 for demo tape, or SASE for info only to:
SoundVault, PO Box 2932
Santa Clara, CA 95055-2932

MIDITRON—The easy way to preview sequences from the leading vendors, artists, and composers. New releases, original compositions & special promotions. **MIDITRON** 24-hour line: (614) 888-0802. Info: Data Assist, Inc., 659-H Lakeview Plaza Blvd., Columbus, OH 43085. Phone: (614) 888-8088.

STONEHOUSE STUDIOS
MIDI software, all major brands. Notation, Educational, Composing, Sequencing, Digital/Audio Programs, Sound Cards, MIDI Interfaces. Visa/MC/Amex/Dis.
(800) 646-4362

Composers, Improvisers...The only software that gives you control of your music ideas is "LICKS" for the Macintosh, $30 + $5 S&H. Joining **"Sightreading MasterTutor"** and others, all satisfaction guaranteed. Info free; demo $5. **SoundWise**, PO Box 3573, Portland, OR 97208-3573. Tel. (503) 626-8104.

We're celebrating our 10th year in business! Save $200 on the **2000-Series** for the **K2000**. 106 disks only $295. Our 81-disk **Elite Series** for the **S550/330, W30** now only $250. Professionally progammed, ready to play. Free U.S. shipping/lifetime replacement. Send check/MO to: **PATCHMAN MUSIC**, 2043 Mars Ave., Lakewood, OH 44107. (216) 221-8887.

CLASSIC ROCK AND BLUES SEQUENCES BY PETER SOLLEY. Producer for Motorhead, Nugent, Romantics, etc. Keyboardist for Clapton, Whitesnake, and many others. **We are the best**. Call (305) 979-8206 or fax (305) 979-0943 for free 15-minute demo and song list. Visa/MC accepted.

COMPUTER MUSIC PRODUCTS for IBM/PC musicians. Great prices & selection of popular MIDI software/hardware. Call for FREE catalog. **(813) 751-1199.** Beginners welcome!

SAMPLER OWNERS. We have the sounds you need! CD-ROMs, Audio CDs, Floppy Disks, Memory, Hard Drives. **CALL GREYTSOUNDS (800) 266-3475**. 501 4th Street SE, Bandon-by-the-Sea, OR 97411. (503) 347-4700 • FAX (503) 347-4163.

VL1/VL1-m Owners! Supercharge your VL with 2 all-new, breath-controlled soundbanks. The most responsive, expressive sounds available! **Volume 1:** Trumpets, saxes, trombones, exotic flutes, harmonicas, guitars, basses, violins, big bands, Brecker-style leads, more! **Volume 2:** Killer layers, analog synths, Moog basses, flutes, guitars, brasses, more! 64 voices each. $41.95 each. $72.95 for both. Send check/MO to: **PATCHMAN MUSIC**, 2043 Mars Ave., Lakewood, OH 44107. 216-221-8887.

DANGEROUS SOUNDS! The best patches and samples for **every Ensoniq keyboard**, from the KT AND TS to the Mirage. Hot **Kurzweil K2000** samples, slamming new patches plus factory libraries for **Casio VZ** synthesizers. Free catalog! Syntaur Productions, (800) 334-1288; (713) 682-1960.

MidiGig's GM/GS MIDIfile Special. 169 songs for $169. Unbelieveable but true! Our customers say "MidiGig Is the Best". Send $3 for DOS or Mac demo file. Call A Better Production, Box 8544, Riverside, CA 92515-8544. (909) 351-4528.

Increase productivity with **BANDMAN!** Professional band mgmt PC-software system. Does promos, set, calendar, schedules, invoices, mailings, contacts, inventory & MORE! Only $149.99! Includes P&H. MS-DOS/4M RAM/3.5" Floppy. Send check or money order to: BandMan! c/o Pandium Software, 42 Carroll Rd., Pasadena, MD 21122. Tel. (410) 647-8386.

The **WORLD MUSIC MENU** for Windows or Macintosh. Instantly transform your synthesizer to play over 100 ancient & modern scales. From Greece to Tibet, from Bali to the Blues, immerse yourself in an exciting new universe of sound. Free Play Productions, (310) 459-8614; fax: (310) 459-8801.

Hot Tropical Music: MidiFiles, print sheets, custom sequences & arrangements. Free catalog. Free S&H worldwide. GM/SG. **Midi-Center**, Suite 344, PO Box 4956, Caguas, PR 00726-4956. (809) 746-4852; fax: (809) 744-7772.

MIDI Sequences
All types of useful arrangements for Today's Musician! Available for PC, Mac, Atari & various workstation environments!
• Techno • House • Industrial
• Rave • Disco • Retro
• Pop • Hip-Hop • Funk
• Rap • Rock • Alternative
Dance Beats™ 680 fast & hard drum patterns. DB001 & DB002.....$39.95 EA
Groove Master™ 60 driving 4-bar basslines. GM003 & GM004.....$39.95 EA
Pop Piano™ 50 sixteen-beat chord progressions. PP005 & PP006.....$39.95 EA
Cultural Collection™ 400 drum patterns with *Country, Blues, Jazz, Standards, Ballads* & more. CC007.....$29.95
The Arpeggiator™ 30 slashing, arpeggiated lix. TA008.....$19.95
For more info and a free catalog
Call (713) 852-0444 Today!
Ask About Our Package Prices
CH/MO CC/COD
Big City Productions
Box 263 • Humble, TX 77347

ENSONIQ OWNERS: Convert Sequences to/from **Standard MIDI files** on IBM-PCs. Each package TS-10/12, ASR-10, EPS/EPS16, VFX-SD/SD-1, SQ-80, SQ-1/2, KS32, 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.

DPM 3/4, EIII, SP S1000 Owners!! We 've got hundreds of killer INFINITY looped sample disks! Memory upgrades, too! VISA/MC. Catalog: Sound Engineering, Attn: E10, PO Box 945, Frazer, PA 19355.
(610) 519-WAVE

VISA MasterCard
twiddly-bits
" Killer riffs played by professional musicians"
...is how Sound on Sound magazine described Twiddly.Bits

Sequencing authentic 'real instruments' parts from a keyboard is difficult. Actually, impossible for most of us.

Which is why we spent hundreds of studio hours recording people like **Bill Bruford, Steve Hackett, Milton McDonald,** and **Gavin Harrison** playing MIDI versions of their instruments.

Now you have access to this stunning collection of parts, grooves, licks, and fills, as MIDI data, on disk.

Vol. 1 General Instruments
Vol. 2 Gate Effects ($29.95)
Vol. 3 Acoustic & Electric Guitar
Vol. 4 Drums & Percussion
Twiddly. Beats Vol. One Brazilian Rhythms
NEW! Vol 5 Country—*Fiddle, guitar, banjo, piano...from some of the hottest names in country. Must be heard to be believed !*

Books also available-
Keyfax- the world's best selling series on electronic keyboards:
Keyfax-1 sold out Keyfax-3 $19.95
Keyfax-2 $14.95 Keyfax-4 $24.95
Keyfax-5 $24.95

Price: $34.95 Visa/MC/Discovery accepted. Add $4 shipping and handling.
More information or order from
KEYFAX Software,
PO Box 958, Aptos, CA 95001-0958
Call 408-688-4505, or FAX 408-689-0102

CLASSIFIEDS

Harmonize your melody. Unique PC S/W searches for the right chords in any key. Requires VGA & MPU401 compatible I/F. Writes SMF. $29.95 + $3 S&H. VHV, 24690 Saddle Peak Rd., Malibu, CA 90265. (310) 456-2207.

BAND-IN-A-BOX IMPROVEMENT PRODUCTS: rated "A good buy" by *Electronic Musician* magazine! Power-User Styles, Fake Disks, and More! Tune 1,000 brand GenMIDI **SEQUENCES** too! **FREE** info! Send legible address today: Norton Music & Fun, Box 13149, Ft. Pierce, FL 34979. Voicemail/Fax (407)467-2420; notesnortn@aol.com

SOFTWARE BLOWOUT! Cakewalk Pro OR Finale Academic—$199.95!! Discount Sequences by Tune 1000, Turbo Music, Tran Tracks. All the leading software, hardware. Huge classic guitar MIDI library. Visa/MC. Call **(800) 787-6434**. FREE catalog. Dept. E, **MIDI Classics®**, 81 Latimer Lane, Weatogue, CT 06089.

COUNTRY SEQUENCES
We do country best CUZ country's all we do. **C.J. MIDI PRODUCTIONS**, 24 Hinkleyville Rd., Spencerport, NY 14559.
(716) 352-5493

MISCELLANEOUS

MAKE A FORTUNE IN THE JINGLE BUSINESS ! ! !

Call 1-800-827-1366 for a FREE RECORDED MESSAGE 24 HOURS and learn how. I'm a 17 year veteran with jingles in every state. My complete jingle course shows you exactly how to do the same. Part-or-full time, locally-or-nationally. CALL NOW This information will save you years of trial and error. MAKE MONEY WITH YOUR MUSIC.

CLASSIFIED ADS DEADLINES
OCTOBER 1ST—DECEMBER '95 ISSUE
NOVEMBER 1ST—JANUARY '96 ISSUE
Call Robin Boyce—800-544-5530

EM CLASSIFIEDS WORK FOR YOU

Text rate:	**$8 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. $56 MINIMUM CHARGE for each ad placed.**
Enhancements:	$10 black border, $15 for a gray-screened background, $25 for a reverse. $25 for Post Office box service. Charges are based on a per-insertion basis.
Display rate:	$100 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:	$25 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.
Send coupon & payment to:	**Electronic Musician Classifieds:** Attn: Robin Boyce, 6400 Hollis St., #12, Emeryville, CA 94608, tel. **(800) 544-5530** or (510) 653-3307; fax (510) 653-8171.
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 (check one):

- ☐ ACOUSTIC CONSULTING
- ☐ EMPLOYMENT
- ☐ EQUIPMENT FOR SALE
- ☐ INSTRUCTION & SCHOOL
- ☐ PARTS & ACCESSORIES
- ☐ PUBLICATIONS
- ☐ RECORDING SERVICES & REPAIRS
- ☐ RECORDS, TAPES & CDS
- ☐ WANTED TO BUY
- ☐ SOFTWARE & PATCHES
- ☐ MISCELLANEOUS

ATTACH YOUR CLASSIFED 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 ($100 per inch)** $_____

_____ **Lines @ $8** $_____

(seven-line minimum)

_____ **Bold @ $0.50 additional** $_____

_____ **Border @ $10** $_____

_____ **Reverse @ $25** $_____

_____ **Screen @ $15** $_____

_____ **Blind PO box @ $25** $_____

_____ **Special Saver rate =** **$ 25**

TOTAL PAYMENT INCLUDED $ _____

☐ Visa ☐ MC ☐ AmEx

☐ Check/Money Order # _____

Card # _____

Exp. _____

TECH PAGE

Online Radio

Surf the Net and hear indie bands via modem.

By Scott Wilkinson

The Internet offers musicians of all kinds unprecedented opportunities. Among the most intriguing possibilities is real-time audio broadcasting to anyone who visits a certain World Wide Web site. A number of college radio stations have already started to rebroadcast their on-air programming on the Internet, including WXYC at the University of North Carolina, KUGS at Western Washington University, and WJHK at the University of Kansas, Lawrence.

Unfortunately, these broadcasts require special high-bandwidth phone lines and modems, to which relatively few people have access. However, a new endeavor seems destined to change that. Radio HK was developed by the Hajjar/Kaufman New Media Lab as the first real-time audio broadcaster exclusively on the Internet. Specializing in independent rock bands and alternative music, Radio HK broadcasts 24 hours a day with a monaural signal most listeners find roughly comparable with the sound quality of AM radios.

Radio HK uses two methods of broadcasting. Their initial implementation uses *CU-SeeMe,* a free teleconferencing software program developed at Cornell University. (The aforementioned college-radio rebroadcasts also use this system.) This software requires a Macintosh or UNIX-based computer and an Internet connection that is faster than conventional telephone lines; ISDN bandwidth of 56 Kbps is the minimum. In this case, the programming is "live," just like on conventional radio. To download the *CU-SeeMe* client software, use your Web browser to access Cornell's site at ftp://cu-seeme.cornell.edu/pub/video.

More recently, Radio HK added a second delivery system called *RealAudio* by Progressive Network, a Seattle-based server-software developer. This software allows anyone with a Macintosh or a Windows machine and sound card, a 14.4 Kbps s modem, and Web access to tune in (see **Fig. 1**). To download the *RealAudio* client software, visit the Radio HK Web site (http://www.radiohk.com/radio) or Progressive Network's Web site (http://www.realaudio.com/). Progressive Network also offers several other *RealAudio*-based broadcasts from their Web site, including National Public Radio's *Morning Edition* and *All Things Considered,* various programs from ABC News, and selections from Adam Curry's Metaverse.

With *RealAudio,* the Radio HK programming is not continuous. Instead, a listener requests the current program, which is stored on a Web server. There is a single 1-hour program available at any given time that is updated approximately twice a month. You can also assemble a playlist of tunes from around the world in the Indy Listening Bar, which functions like a juke box. The delay between a request and the start of playback is generally less than two seconds.

FIG. 1: The Radio HK home page lets you download *RealAudio* client software and tune in to the world's first Internet-only audio broadcaster.

Once the program begins, compressed audio is sent at a rate of 8 KB per second, which accounts for the AM-like sound quality. Unlike the *CU-SeeMe* approach, the listener has some control over the program, including fast forward and rewind. In addition, listeners can continue to listen as they surf to different Internet sites, download files, and engage in any other online activity.

About 100 people can listen to Radio HK simultaneously, thanks to Hajjar/Kaufman's T1 phone line, which boasts a bandwidth of 1.5 Mbps. In the first twelve weeks of *RealAudio* operation, the site logged more than 4 million "hits," which translates to at least 70,000 listeners. These listeners hail from 6 continents, more than 40 countries, and all 50 states.

As mentioned earlier, Radio HK focuses on independent bands. They even accept recordings from unsigned groups to include in their programs. In addition, they've acquired the first ASCAP license to broadcast on the Internet, which makes more than 3 million titles available. Clearly, this is just the sort of resource that computer-savvy musicians should investigate.

ARE YOU GEARED UP FOR THE NEXT REVOLUTION?

KURZWEIL'S POWERFUL NEW K2500R/S

THE PRODUCTION STATION – Kurzweil introduces its most powerful sampling synthesizers ever: the **K2500R** and **K2500RS**. So powerful and flexible, they let you take a project from conception to completion without leaving the instrument. So advanced, they interface with a myriad of analog and digital formats.

The **K2500** racks combine Kurzweil's proprietary *V.A.S.T.*® synthesis, 48-note polyphony, a new soundset, effects processor, sequencer, and Advanced File Management System (AFMS).

V.A.S.T. POWER – *Variable Architecture Synthesis Technology* offers 60 DSP functions arrangeable in 31 algorithms, allowing you to use different methods of synthesis within the same program.

The DMTi option for direct digital connection to the Alesis ADAT® and Tascam® DA88.

UPGRADABILITY – Optional upgrades for the **K2500R** virtually eliminate the need for outside gear. The *KDFX Digital Effects Option* provides four stereo buses and state-of-the-art DSP. The *DMTi Digital Multitrack Interface* offers four stereo channels of digital format and sample rate conversion in real time. The *Orchestral ROM*, *Contemporary ROM* or *Stereo Digital Piano SoundBlocks* offer a total of 28 MB of Kurzweil's finest sounds.

SAMPLE WHILE PLAY – The *SMP-2 Sampling Option* has digital and optical I/O; its sample-while-play feature lets you record samples while continuing other operations.

SOFTWARE UPGRADES IN A FLASH – With *Flash ROM*, you can upgrade your operating system from floppy disk or via SCSI.

STUNNING ON-BOARD SOUNDS – For the **K2500R**, Kurzweil unveils a stunning new soundset of 200 programs and 100 setups, plus a separate disk containing 1,000 useful programs.

You can also load samples from most formats (DOS, AIFF, .WAV, Roland®, Akai®, Ensoniq®) and process them with *V.A.S.T.*

THE REVOLUTION BEGINS – Want to gear up for the next revolution? What are you waiting for? Visit your local authorized Kurzweil dealer today, and try the **K2500R** for yourself.

- ***48-note polyphony***
- ***Up to 128 MB sample RAM***
- ***Up to 28 MB internal ROM sounds***
- ***Up to 1.26 MB program memory***
- ***32-track sequencer***
- ***28 on-board effects algorithms***
- ***Flexible analog & digital interfacing***
- ***Dual SCSI ports***
- ***Easy upgradability***

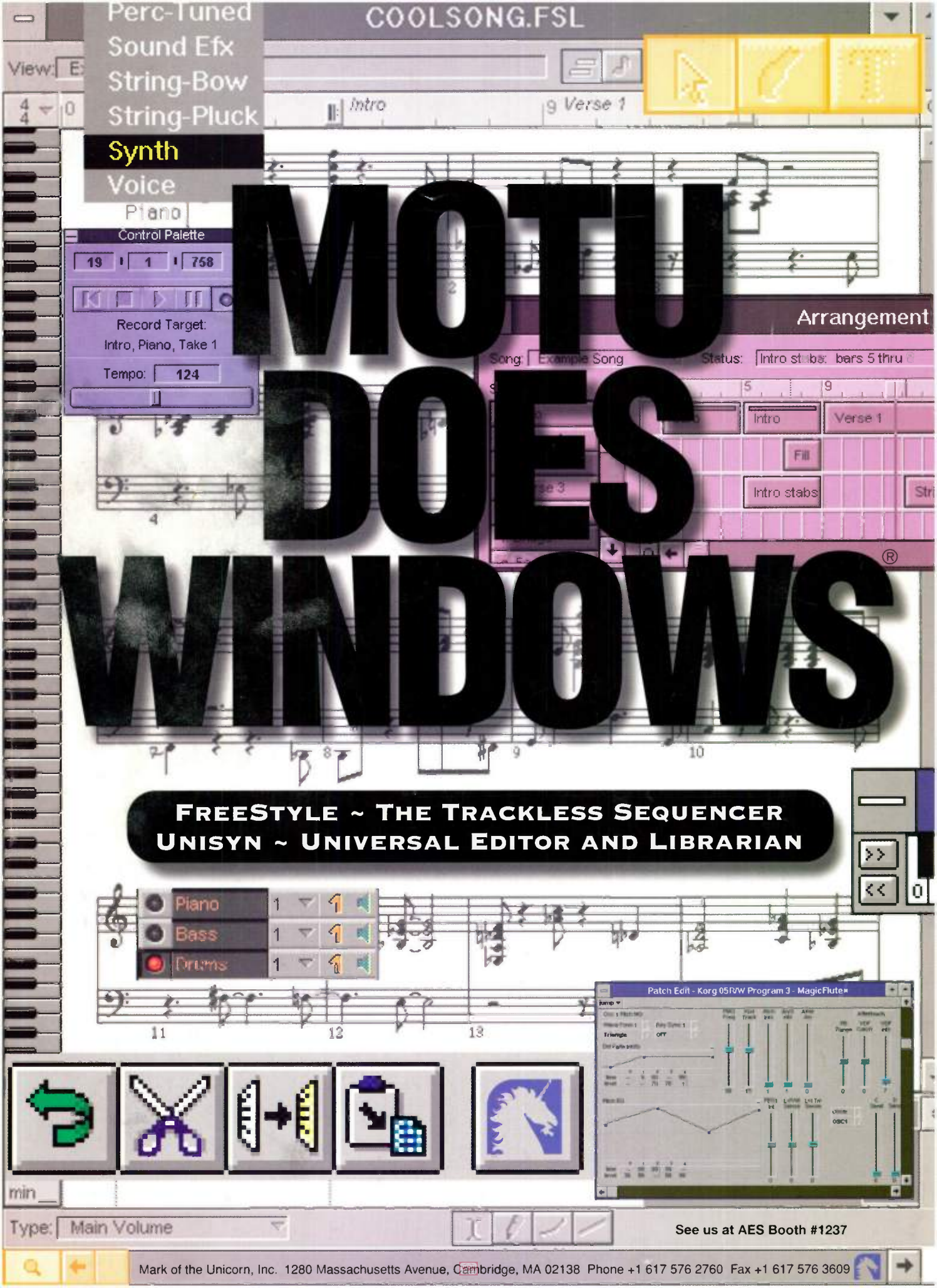
Get a Free Copy of the "Players of the World" poster at your Authorized Kurzweil Dealer.

KURZWEIL®
Music Systems
Kurzweil is a product line of Young Chang

PURE INSPIRATION

See us at AES Booth #850

13336 Alondra Blvd. Cerritos, CA 90703 Tel: 310/926-3200 Fax: 310/404-0748 E-Mail: kurzweil@aol.com Web: http://www.musicpro.com/kurzweil/


COOLSONG.FSL
Perc-Tuned
Sound Efx
String-Bow
String-Pluck
Synth
Voice
Piano
Control Palette
Record Target:
Intro, Piano, Take 1
Tempo: 124
Arrangement
Song: Example Song
Status: Intro stabs: bars 5 thru
Intro
Verse 1
Fill
Intro stabs
MOTU
DOES
WINDOWS
®
FREESTYLE ~ THE TRACKLESS SEQUENCER
UNISYN ~ UNIVERSAL EDITOR AND LIBRARIAN
Piano
Bass
Drums
Patch Edit - Korg 05R/W Program 3 - MagicFlute
Type: Main Volume
See us at AES Booth #1237
Mark of the Unicorn, Inc. 1280 Massachusetts Avenue, Cambridge, MA 02138 Phone +1 617 576 2760 Fax +1 617 576 3609