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August 1999

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A PERFECT 10

Which digital mixer ? is right for you

Mark Hudson records music the old-fashioned way

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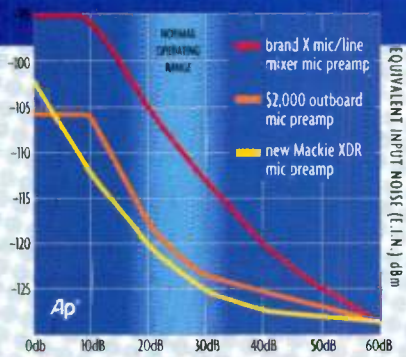
NEW VLZ PRO

THE FIRST MIXERS WITH EXPENS

Two years in the making, XDR™ Extended Dynamic Range the pristine sonics and awesome specs of high-end ou

If your hype alarm is going off, we can't really blame you.

The proof is in the listening. Visit your nearest Mackie dealer and audition the XDR™ preamp design in our new VLZ PRO Series compact mixers. Use the most expensive microphone they have. Compare XDR to ultra-expensive outboard mic preamps. Compare it to our compact mixer competition. Bring your golden-eared audiophile friends. We think you'll be amazed. We honestly believe that you've never before heard a mic preamp this good.



A new standard for low noise in the critical +20 to +30dB operating range.

If money is no object, don't read any farther.

If you can afford $1000 a channel for outboard "audiophile" mic preamplifiers, DO IT! Because no matter how much you spend on a microphone, its ultimate performance depends on how it interacts with the preamp it's plugged into.

Yes. We openly admit it: Many high-end mic preamps can effortlessly amplify the slightest sonic nuance, creating an aural panorama that's breathtakingly realistic, excitingly vivid and truly 3-dimensional in scope. For years, they've provided fidelity that just hasn't been possible with the "stock" mic preamps built into mixing consoles.

Until now.

A massive R&D initiative.

We can confidently say that no other company in the world has spent the sheer number of engineering hours —and $250,000 in R&D costs that we just did on a single new microphone preamplifier design.

The XDR team started with blank paper, concerned only with matching or exceeding the performance of $500 to $2000-per-channel esoteric preamps. They went through hundreds of iterations and revs and spent countless hours subjectively listening (and arguing). They started all over again several times. They scoured the world for rare parts. Then they spent more time critically listening and evaluating the

NEW!

1604-VLZ PRO

16x4x2 · 16 XDR™ preamps

60mm faders • 16 mono chs. • 4 sub buses • main L/R • 3-band EQ with sweepable midrange (12kHz & 80Hz shelving, 100Hz-8kHz mid) • 18dB/oct.@ 75Hz low cut • 6 aux sends per ch. • Constant Loudness pan controls • 4 stereo aux returns • RCA tape inputs & outputs • 16 channel inserts, 16 high-headroom line inputs • 8 direct outs • TRS balanced outputs • Switchable AFL/PFL Solo • Ctl Room/Phones matrix with Assign to Main Mix & separate outputs • Ctl Rm/Phone level control • 12-LED metering plus Level Set LED & RUDE Solo light • Aux 1 & 2 Pre/Post • Aux Send master section w/level controls • Solo buttons with LEDs • Stereo Aux Return assign section with EFX to Monitor & Main/Submix assign • built-in power supply • solid steel main chassis • BNC lamp socket • Rotatable I/O pod allows 5 different physical configurations

NEW!

1402-VLZ PRO

14x2x1 · 6 XDR™ preamps

60mm faders • 6 mono & 4 stereo chs. • 3-band EQ @ 12kHz, 2.5kHz & 80Hz • 18dB/oct.@ 75Hz low cut • 2 aux sends per ch. • Constant Loudness pan controls • 2 stereo aux returns • RCA tape inputs & outputs • 6 channel inserts, 6high-headroom line inputs • XLR & TRS balanced outputs • switchable +4/mic level output • ALT 3-4 stereo bus • Switchable AFL/PFL Solo • Ctl Room/Phone matrix with Assign to Main Mix & separate outputs • Ctl Rm/Phone level control • 12-LED metering plus Level Set LED & RUDE Solo light • Aux 1 Pre/Post • EFX to Monitor • sealed rotary controls • built-in power supply • solid steel chassis

NEW!

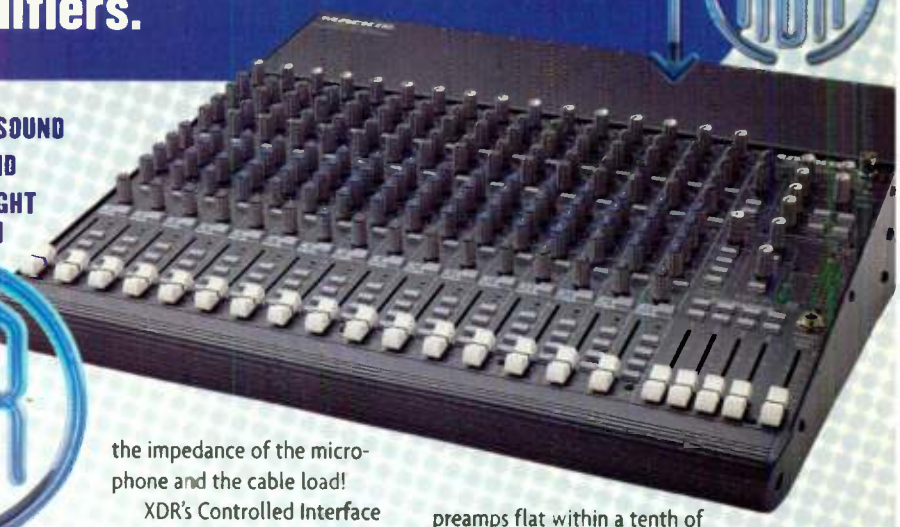
1202-VLZ PRO

12x2x1 · 4 XDR™ preamps

4 mono & 4 stereo chs. • 3-band EQ @ 12kHz, 2.5kHz & 80Hz • 18dB/oct.@ 75Hz low cut filter • 2 aux sends per ch. • Constant Loudness pan controls • 2 stereo aux returns • RCA tape inputs & outputs • 4 channel inserts, 4 high-headroom line inputs • XLR & TRS balanced outputs • switchable +4/mic level output • ALT 3-4 stereo bus • Ctl Room/Phones matrix with Assign to Main Mix & separate outputs • Ctl Rm/Phone level control • 12-LED metering plus RUDE Solo light • Aux 1 Pre/Post • EFX to Monitor • sealed rotary controls • built-in power supply • solid steel chassis

IVE ESOTERIC MIC PREAMP SOUND

mic preamp circuitry gives our new VLZ PRO™ Series board microphone preamplifiers.



WARM, DETAILED SOUND
0.0007% THD
NEAR DC-TO-LIGHT BANDWIDTH

OVER 130dB DYNAMIC RANGE FOR 24-BIT, 196kHz SAMPLING RATE INPUTS

ULTRA-LOW IM DISTORTION & E.I.N. AT NORMAL OPERATING LEVELS

IMPEDANCE INDEPENDENT

BEST RF REJECTION OF ANY MIXER AVAILABLE

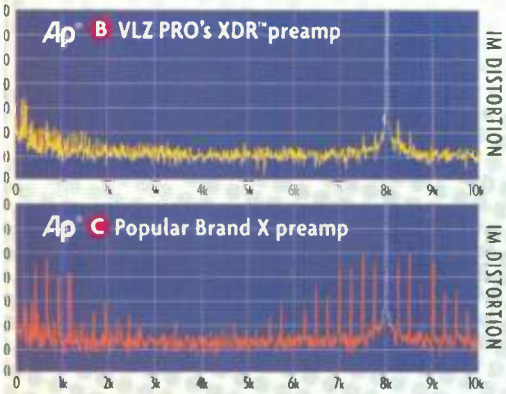
design with every high-end microphone you can think of. Then they brought in veteran recording and live sound engineers for more exhaustive listening tests.

What we ultimately ended up with is not just an awesome sounding design. XDR is also **a)** highly resistant to damage caused by "hot patching" (caused by routing a phantom powered mic through a patch bay); **b)** remarkably independent of cable-induced impedance variations; and **c)** able to reject extremely high RF levels without compromising high frequency response.

Rejecting RFI without tuning out sound quality.

Because a mic preamp must amplify faint one millivolt input signals up to a thousandfold (60dB), its rectification components can also pick up radio frequency interference (RFI) from AM and FM stations, cell phones and pager transmitters—even microwave ovens—and amplify them to audible levels.

We assaulted RFI on three fronts. First, we incorporate bifilar wound DC pulse transformers with high permeability cores that reject RFI but don't compromise audible high frequency response. Second, we carefully-matched high-precision components for critical areas of the XDR preamplifier. Third, we direct-coupled the circuit from input to output and used pole-zero-cancellation constant current biasing (which also avoids increased intermodulation distortion at high signal levels).



XDR vs. Brand X FFT analysis of mic preamp Intermodulation Distortion. Mixer trims at 30dB, 0dB at inserts. The white spike at 8kHz is the fundamental tone used to "generate" the surrounding distortion artifacts—which the Brand X mic preamp has far more of.

Bottom line for the non-technical: Our new VLZ PRO Series has the best RFI rejection of any compact mixers in the world. Period.

Controlled Interface Input Impedance.

If a mic preamp isn't designed right, it will actually sound different depending on the impedance of the microphone and the cable load!

XDR's Controlled Interface Input Impedance system accepts an enormous range of impedances without compromising frequency response. Whether the mic/cable load is 50 ohms, 150 ohms or 600 ohms, XDR mic preamp frequency response is down less than one tenth of a dB at 20Hz and 20kHz!

Ultra-low noise at "Real World" gain settings.

Many mixers that tout low E.I.N. (Equivalent Input Noise) specs can't deliver that performance at normal +20 to +30dB gain settings. Graph A on the other page charts E.I.N. versus gain level for our new VLZ PRO Series vs. a major competitor's mic/line mixer preamps and a "status" outboard mic preamp retailing for about $2,000. As you can see, our XDR design maintains lower noise levels in the critical +20 to +30 gain range than either competitor.

There's still more:

- **0.0007% Total Harmonic Distortion.** The lowest ever in any compact mixer.
- **Flat response.** Not only are XDR mic preamps flat within a tenth of a dB across the bandwidth of any known microphone, but are also only 3dB down at 10Hz and 192kHz!
- **116dB CMRR** 20Hz to 200kHz and above.
- **Super-low intermodulation distortion** at very high operating levels (charts B&C at left) thanks to instrumentation-style balanced differential architecture, linear biasing and use of DC-coupled pole-zero-cancellation constant current that frees the mic preamp from power supply fluxuations.

We could go on and on this way. But like we said at the start of this ad...

Hearing is believing.

Visit your nearest Mackie Dealer. Select a really high-quality condenser mic and try out the new 1604-VLZ PRO, 1402-VLZ PRO and 1202-VLZ PRO Think of them as expensive esoteric mic preamps... with really excellent compact mixers attached.

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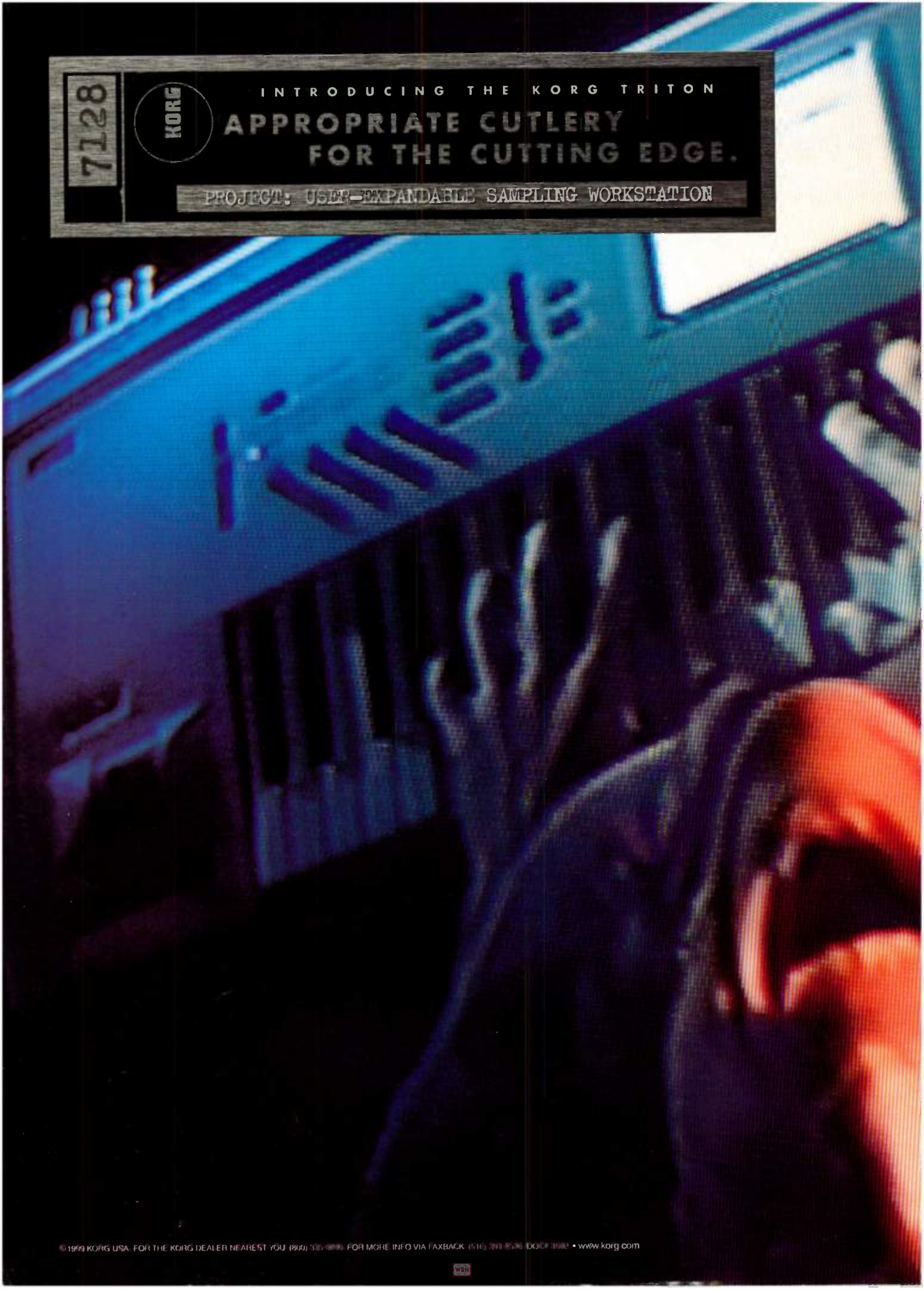
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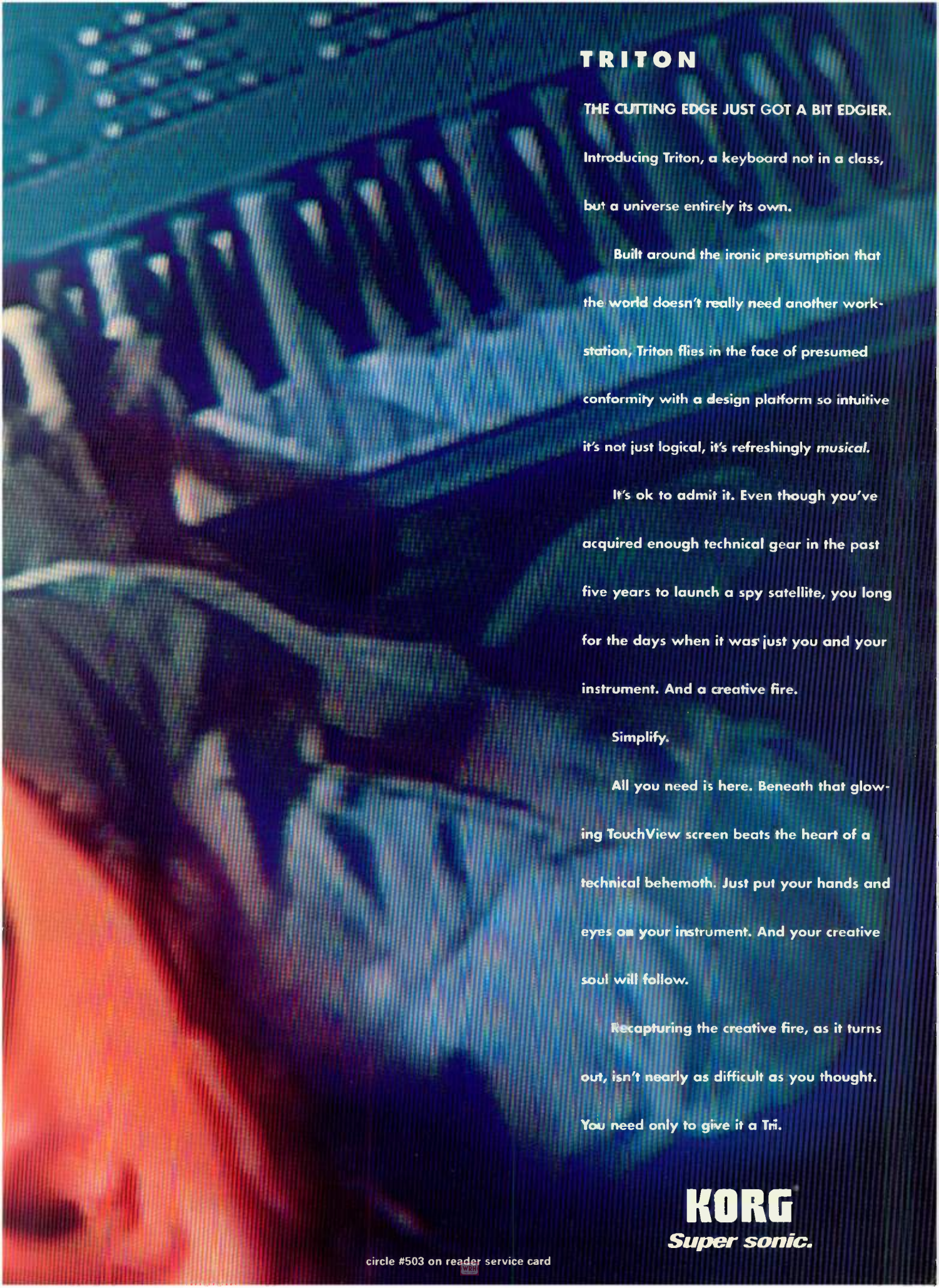

TRITON

THE CUTTING EDGE JUST GOT A BIT EDGIER.

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Built around the ironic presumption that the world doesn't really need another workstation, Triton flies in the face of presumed conformity with a design platform so intuitive it's not just logical, it's refreshingly *musical.*

It's ok to admit it. Even though you've acquired enough technical gear in the past five years to launch a spy satellite, you long for the days when it was just you and your instrument. And a creative fire.

Simplify.

All you need is here. Beneath that glowing TouchView screen beats the heart of a technical behemoth. Just put your hands and eyes on your instrument. And your creative soul will follow.

Recapturing the creative fire, as it turns out, isn't nearly as difficult as you thought. You need only to give it a Tri.

KORG

Super sonic.

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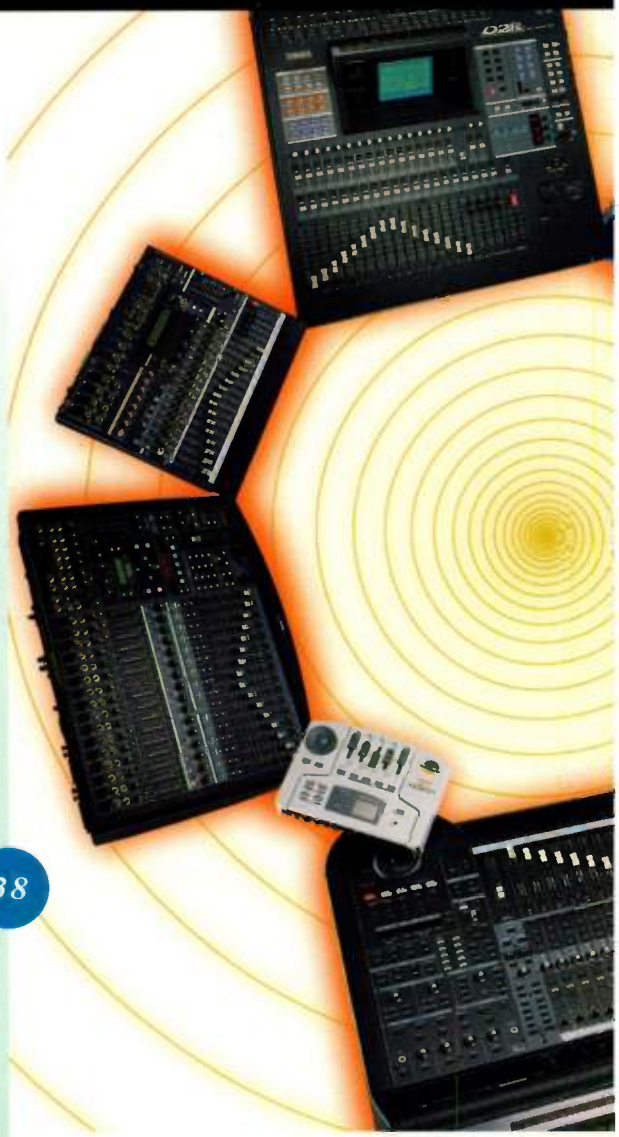
I N S I

FEATURES



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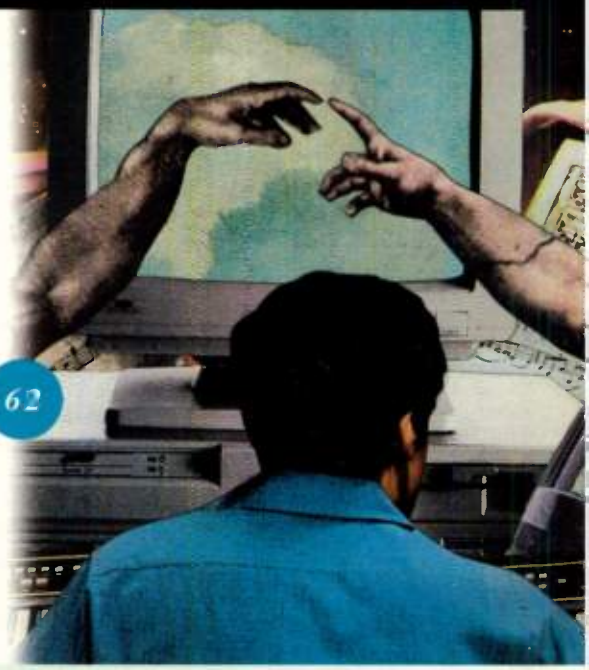
DEPARTMENTS

Electronic Musician
AUGUST 1999 VOL. 15, NO. 8

COLUMNS

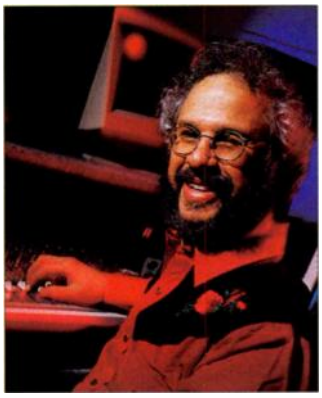


REVIEWS

Cover: Photography by Robert Perry

The Chairman of the Boards

Why I'm saving my pennies for a digital heart transplant.



ANTHONY PIDGEON

During the past year, I redesigned my entire studio. In order to reduce room noise and clutter, I built an equipment closet, pulled and rerouted every cable, reconfigured the racks, added new signal processors, modified my computer, improved the AC system, and much more. This huge project took quite a while—especially because I had to fit it in around **EM**'s hectic publishing schedule.

A large part of the redesign was figuring out the best way to rearrange and rewire the myriad gadgets I have accumulated in my studio over the years. In the process, I also realized how frustrated I am with my aging, undersized mixer, which forces me to use several submixers and patch bays to route my many synthesizers, samplers, and processors.

I was tempted to cut down to a few key pieces and get on with making music. After all, you don't always need racks of gear; Mark Hudson (interviewed in "A Room with a Vibe" on p. 72) once recorded a commercial Ozzy Osborne album on a Tascam Portastudio. But as I started reviewing my instruments and recording gear, I realized I couldn't bear to part with most of it.

The solution? Unfortunately for my bank account, what I need is a digital mixing console that has lots of analog inputs and Lightpipe I/O. I could dump the old mixer and submixers and eliminate some outboard gear, which would reduce the amount of cabling. At the same time I could more freely transfer tracks to and from my instruments, processors, MDM, and computer. Furthermore, I already use a computer to control as much of my rig as possible, and most digital mixers would let me automate my entire mix from a sequencer, including tape tracks, EQ, dynamics processing, and perhaps some multi-effects. That's a sweet prospect. So I'm saving for a digital console and contemplating the inevitable: another (much less ambitious) studio redesign.

But which digital console? As of this writing, there are a dozen models for personal-studio and live-performance applications, though a couple are apparently being phased out. Several new ones should be available by the time you read this. Each mixing board has a distinctive "personality" in terms of user interface, features, and connectivity, and the price range is amazing. I've test-driven a few models, but I haven't had extensive experience with them yet.

Therefore, in a selfish display of raw power, I dubbed Associate Editor Jeff Casey "Chairman of the Boards" and assigned him to test ten digital mixing boards, which was no small task. Thanks to his research (see "A Perfect Ten" on p. 38), you and I are in a better position to figure out which consoles would best suit our needs.

Digital mixers are no longer the wave of the future; they're happening now. I am very excited about what they can do for personal studio owners and performing musicians alike. If you aren't fired up about them yet, I think you will be after you have read the Chairman's report.

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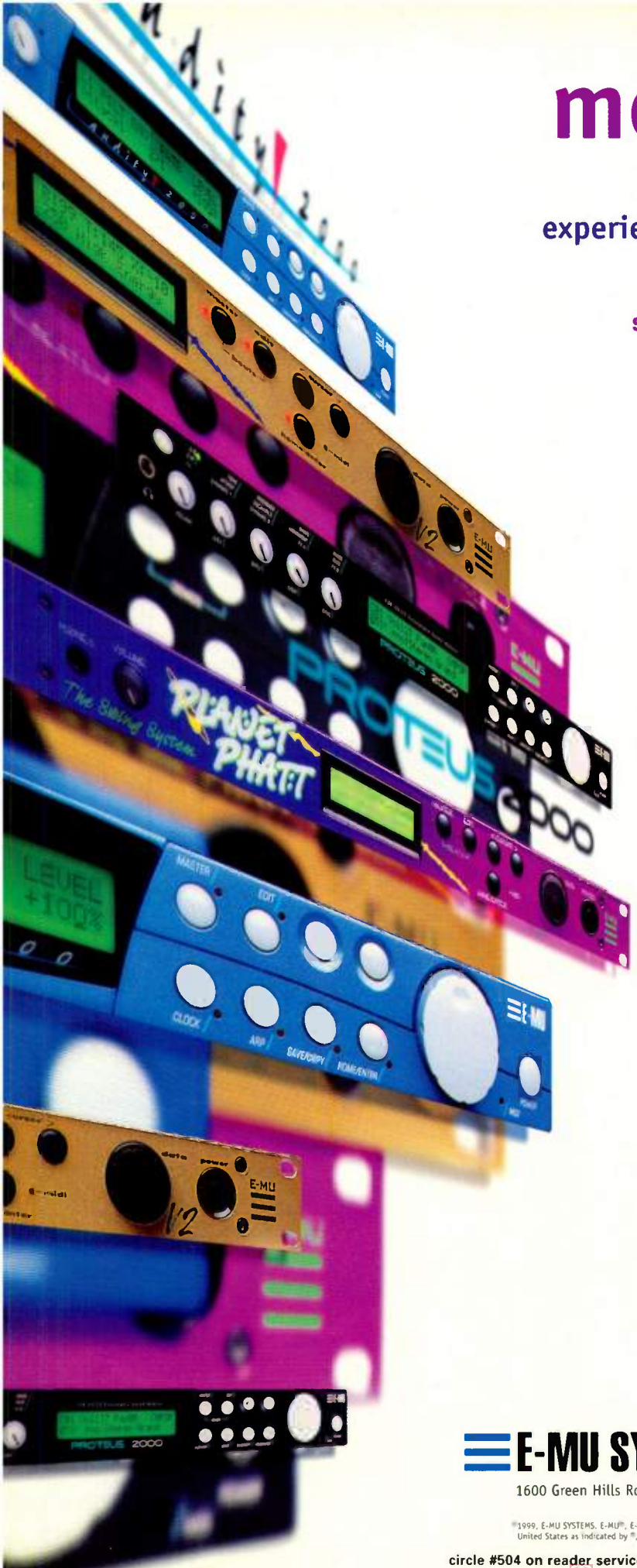

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CANNIBALISM

Hats off to Steve Oppenheimer for saying something that needed to be said ("Front Page: Walk the Plank!" June 1999). Sometimes editors go off on such abstract diatribes that I'm left wondering what they were talking about. Not this time.

I admit that I'm one of those who finds sampling about as artistically gratifying as painting by numbers. However, the problem of outright DSP-software theft is a phenomenon with no gray area. I hope those who participate in this form of digital cannibalism were given, at the very least, some food for thought.

Dann Glenn
fretless@dannglenn.com

HELP YOURSELF

I write concerning Steve O's editorial "Walk the Plank!" I've been using the same digital audio sequencer since 1994 and have bought every upgrade since. There are advantages to ownership, like tech support and having the manual.

At one point, a friend asked to "borrow" my installer CD, and I very reluctantly obliged. When he asked to photocopy the manual I flew off the handle—and did *not* oblige! Unfortunately, our friendship did suffer as a result, and perhaps that is often the reason why bootlegging occurs. As both a programmer and musician, I understand the intellectual-property issues, but I didn't want to spoil a friendship over something that my friend considered "silly"—even though, in my mind, it's serious. If everyone gave away their software, companies wouldn't be able to afford to make the upgrades, and we'd all lose out. Since that incident, my commendations of that product have convinced several other friends to actually buy it, and I have not repeated my mistake. Now my friends get those $20 upgrades, too.

The question that's been bugging me for a long time is where to draw the line. In the aforementioned case, it's obvious. Looping five seconds of a song is obvious. Taking spoken-word samples off the radio, pitch/time shifting them, applying heavy effects, and splicing them is not so obvious. Nothing I use ends up anywhere near a semblance of the original recording by the time I'm done, but I worry about it. I only *lose* money on this (and plenty of it), and I don't keep track of the sources of my samples (I find them at random), so I'm not sure what I should be doing.

Xindreh
xindreh@cerebral-link.com

Xindreh—We consulted with San Francisco Bay Area music-business attorney Michael Aczon, who replied:

As with any legal issue, there are three approaches to consider: theoretical, practical, and ethical. I'll take these one at a time. When sampling a record, the copyright of the composition and the copyright of the master recording are both protected by law and require permission for reuse. If you sample this material without permission and sell a work that uses it—whether you make a profit or not—you are in violation of these copyrights.

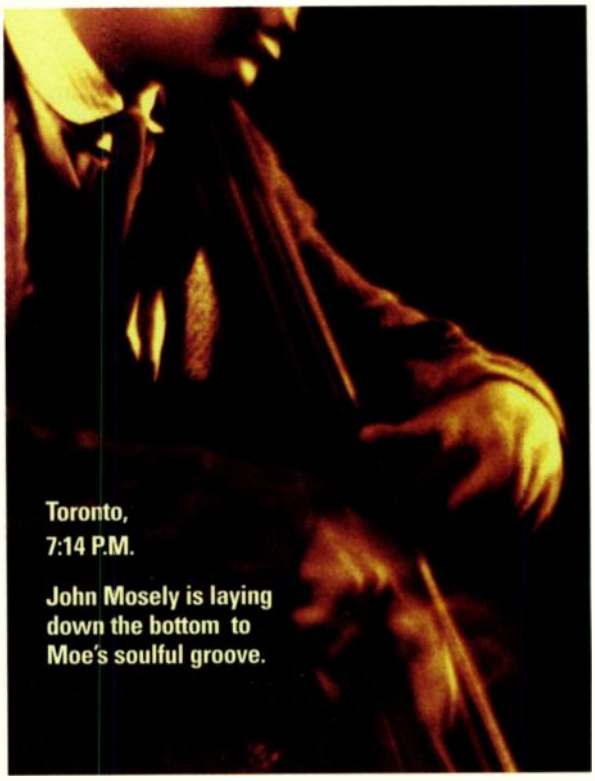
When sampling off of the radio or television, you have to consider the rights to the copyrighted material of the broadcast itself, as well as the rights to privacy and publicity, which are owned by the person whose voice you are sampling. When you make changes to broadcast material, you violate the copyright holder's exclusive right to make such changes and the individual's rights to protect the use of their name, likeness, and voice.

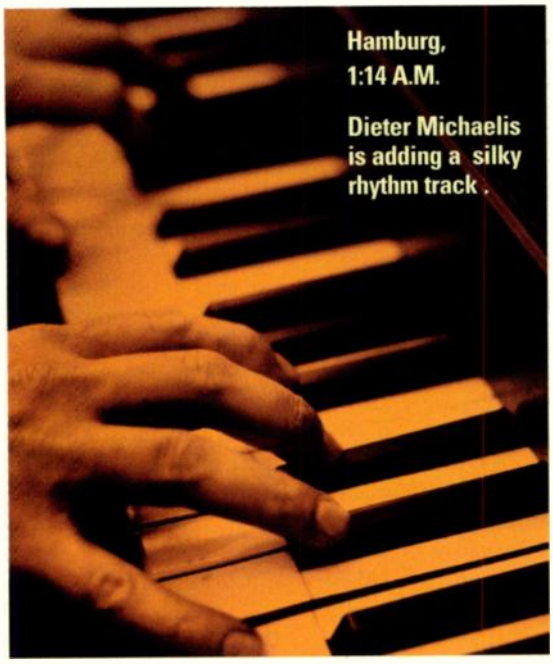
On the practical side, every time you knowingly violate a law, you assume the risk that you must pay the price of the violation. I could tell you that driving four blocks from your house without your driver's license is in violation of the law, but after weighing the risk of the offense, the potential punishment, and the possibility of getting caught, you still might do it. If music that includes pirated material is neither broadcast nor sold in stores, the violations might go unnoticed.

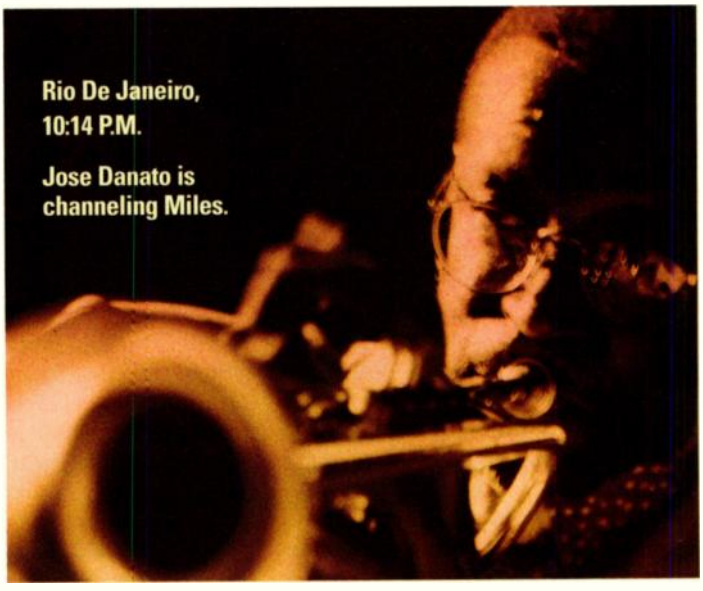
That brings us to ethics. If intellectual-property rights were not enforced, musicians and artists would simply be servants of the general population, unable to be paid based on the sale of their work. When rent time or Christmas comes around, the loss of a couple hundred bucks to you, an independent musician, may be no less a burden than the loss of a couple million bucks to a superstar artist. All musicians would do well to respect and support each other in these matters, rather than take what they please without permission.

TOOT YOUR HORN

Thanks for reviewing the WX5 and VL70-m (June 1999). In my opinion it is the best combination Yamaha has built to date. I'm a saxophonist and have been playing WX-series wind controllers since the WX7 came out in the '80s. I also played the WX11 for years and was one of the first people in the U.S. to get a WX5. I gig with the WX5 two to seven nights per week, using it on about 80 percent of the songs I do on each gig. I have put in a great many hours on the







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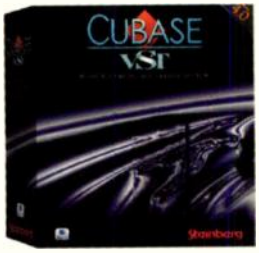

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WX5 and have come to know it well.

Scott Wilkinson did a fine job of reviewing the WX5 and the VL70-m. However, some of the "cons" that Mr. Wilkinson reported were due to his limited understanding of the instrument and the fact that he comes from a recorder background. I don't dispute Mr. Wilkinson's skills as a reviewer or a player—some of these things are only evident after many hours of playing the instrument, and others are sax-centric. I would like to add the following notes to his otherwise fine review.

First of all, there *is* a low $C^{\#}$ key: it is the same as the $G^{\#}$ key. On the sax, the low $C^{\#}$ and $G^{\#}$ keys are physically coupled so that, when you press the $C^{\#}$ key, it also presses the $G^{\#}$ key. Yamaha simply combined the two as one key, which makes changing from the $G^{\#}$ to the $C^{\#}$ very efficient.

Regarding the octave keys, I, too, hated them at first. I played more than my share of wrong octaves, and my thumb got sore, too. However, after a few months of playing, the octave keys just started working for me. Before I sold my remaining WX11s and WX7, I played them to make sure they were in good working order. To my surprise, the old octave-key arrangement was terribly slow and awkward compared to the WX5's system. It takes a couple of months to get used to, but the new octave-key arrangement is a great improvement over the old one. Instant gratification? No. True love? Yes.

I, for one, am glad there is no automatic power-off function. If I am playing sax on a song and want to pick up the WX5 to insert a trumpet solo, the last thing I want is a WX5 that is sleeping. Even a microsecond to wake it up is much too long. There is a manual on/off switch for battery users; it shouldn't take long to get into the habit of using it. I guess Yamaha could have made an either/or function and we both would have been happy, but that might have been too expensive.

One more thing: the Fast fingering mode rules. If your technique isn't up to snuff, it's time to practice. The slow mode adds a bit of latency to the instrument which, to me, is undesirable. I am in complete agreement with most of the other points Mr. Wilkinson made—especially about the lack of a hard case.

I've created a guide containing tips and tricks that I have learned from using the WX5 on the job since last June. **EM** readers can e-mail me for a copy. (It should soon be on my Web site, http://members.aol.com/nortonmidi/index.htm).

Bob "Notes" Norton
Owner, Norton Music
NotesNortn@aol.com

ON THE LEVEL

The June 1999 feature "Shake, Rattle, and Roll Tape" had erroneous information on the construction of the marimba. The article also incorrectly cited four-mallet playing technique as the reason for differences in construction between various mallet instruments.

The marimba, contrary to the author's description, has its sharps and flats placed on a slightly higher level than the naturals. The vibraphone is the only member of the orchestral mallet-percussion family that has all of its bars on the same level; this difference is due to the fact that the vibraphone has a sustain pedal. The pedal controls a spring-loaded damper mechanism, which must contact all of the vibraphone's bars simultaneously in order to work properly.

The four-mallet playing techniques used by percussionists are not hampered in any way by the differences between these instruments; more specifically, the bars do not have to be on the same level in order to play four-note chords that contain sharps or flats.

Despite these errors, I found the article very well written and informative, especially the suggestions regarding mic placement as a solution to prevent transient clipping and digital artifacts.

Mark Griskey
bc853@lafn.org

Author Jim Miller replies: *The two marimbas I sampled both had the sharps and flats on the same level as the naturals. Just to confirm my recollection of these instruments, I consulted the* New Oxford Companion to Music, *published by the Oxford University Press. On page 1,129 of volume 2, the description of the marimba clearly states, "The sharps and naturals are on the same level, to suit playing with four beaters when required." Of course, this is not to say that there are not other marimbas that have been built with raised sharps/flats. As for the limitations of four-mallet playing technique, I based my text on information supplied to me by the owners of the marimbas (and vibraphone) I sampled, which is also supported by the* New Oxford Companion to Music.

POWER TO THE PEOPLE

After reading the article "Seizing Control" (May 1999) about Chris Spheeris's indie success, I felt compelled to respond and praise his decision to save his art and release his recordings independently. I can relate to the problem of a major record label not properly marketing your release. This is very frustrating and discouraging for an artist. Fortunately, artists today have more avenues to release their music without having to sell their souls to the corporate "bottom liners."

Having released two hip-hop albums on a major label, my band is now geared toward releasing a third ourselves, independently. We realize that it's not going to be easy (as Spheeris points out in the article) and will take a lot of self-promoting and hard work, but the outlook is definitely more positive. We bought our own home studio using advance money from the second album and can easily record the third at home. This time around, profits can finally come to us because we've cut out the middle man.

"Seizing Control" is an important article, especially for newer artists. It shows that, with the major advances in affordable digital recording and with marketing tools such as the Internet, artists do have a chance to create freely, make more money than a major-label advance, and get their music heard and sold worldwide. More power to Chris Spheeris and other "uncontrollable" true artists.

O.G.
Los Angeles, CA

ERROR LOG

May 1999, "Seven Studios of Gold," p. 32: We incorrectly reported that Roland VS-840 users could download a free software upgrade from the company's Web site. In fact, the upgrade must be purchased in a $195 package that includes a new drive and labor.

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

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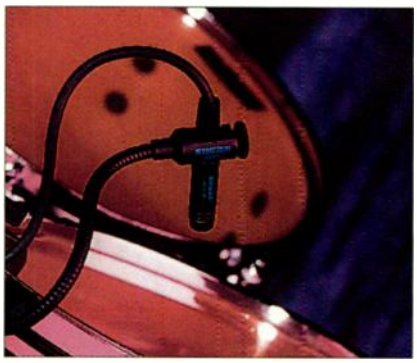
WHAT'S NEW

By the EM Staff



▲ SHURE

Two new drum mics are available from Shure as part of the company's Beta line. The Beta 98D/S ($440) is a miniature condenser specifically designed for miking toms. The polar pattern of the stock Beta 98D/S is supercardioid, but the capsule cartridge can be interchanged with a cardioid-pattern unit ($63.35).

The Beta 91 ($420) is a condenser boundary mic with a flat shape for placement on the padding inside a kick drum, with no extra hardware needed.

The Beta 98D/S and Beta 91 both feature an inline preamp on a detachable cable for quick removal. According to Shure, each mic has a frequency response of 20 Hz to 20 kHz, maximum SPL of 160 dB, and a dynamic range of 125 dB. The Beta 98D/S ships with the A98D gooseneck drum mount and a windscreen. Shure Brothers, Inc.; tel. (800) 25-SHURE or (847) 866-2200; fax (847) 866-2279; e-mail sales@shure.com; Web www.shure.com.

Circle #401 on Reader Service Card

▼ KORG KAOSS PAD

Korg's Kaoss Pad ($350) is a real-time, performance-oriented, X-Y controller and effects processor. Using a front-panel knob, you can select any of more than 60 digital effects, including delays, reverbs, flangers, pitch shifters, and phasers. Several filters are provided for modifying the sound.

The Kaoss Pad takes its name from its control surface, a touch-sensitive X-Y controller pad that you can use to manipulate effects parameters. Each axis controls up to two parameters (the Y axis can control pitch bend, as well), and you can use the diagonal axes to control additional parameters. You can move your finger across the controller to produce smooth parameter changes or tap on it for switching effects, such as turning an effect on and off in time with a rhythm. A Hold switch enables you to freeze the current parameter settings. Because you are able to assign MIDI Control Change numbers for the X and Y axes to transmit, you can also use the Pad purely as a MIDI controller for synths, samplers, and other MIDI devices.

The Kaoss Pad also has a sampler that holds up to five seconds of audio material. You can modify the sampled sound with time-stretch, reverse-playback, and pitch-change functions. All signals are sampled at 48 kHz, and A/D and D/A converters are 18 bit.

Other controls include the Power button, six user-selectable preset buttons, an input-volume knob, an input-source selector switch, and an effect on/off button. The Kaoss Pad has a trim control for mic-input signals and a switch that allows the unit to output either a wet/dry mix or the wet signal only.

With its stereo line and phono inputs (all on RCA connectors) and balanced ¼-inch TRS mic-level input, the Kaoss Pad connects easily to a turntable, CD player, microphone, mixer, or synthesizers. The unit features stereo output on two RCA connectors. Korg USA, Inc.; tel. (516) 333-9100; fax (516) 333-9108; e-mail product_support@korgusa.com; Web www.korg.com.

Circle #402 on Reader Service Card

▼ TRUE AUDIO PRECISION 8

The True Audio Precision 8 ($2,695) comprises eight solid-state mic preamps in a single rackspace unit. According to True Audio, the 8-channel preamp's transformerless design helps the unit deliver detailed, transparent operation. Each channel features a level meter as well as signal-present and overload indicators. Each channel also has a gain knob, which provides 15.5 to 64 dB of boost; a phantom-power switch; and a polarity-reverse switch. A front-panel knob lets you calibrate the channels' peak-hold indicators to your mixing console or recording device.

Channels 1 and 2 have built-in mid-side decoding. Similarly, channels 7 and 8 double as direct-injection inputs for high-impedance instrument level signals. On the right of the front panel are a power switch and a peak-level adjustment knob.

Channels 1 through 6 feature input on balanced XLR connectors; inputs for DI channels 7 and 8 are on balanced/unbalanced XLR/¼-inch combo connectors. Analog output is on eight balanced ¼-inch TRS connectors, and digital output is provided on an 8-channel Tascam TDIF port. The back panel features a ground-lift switch. True Audio rates the Precision 8's frequency response at 1.5 Hz to 500 kHz (+0/-3 dB), equivalent input noise at -132 dB (20 Hz to 30 kHz), THD at 0.0008% (20 Hz to 30 kHz), and crosstalk at less than -130 dB. Neumann USA (distributor); tel. (860) 434-5220; fax (860) 434-3148; e-mail neumlit@neumannusa.com; Web www.neumannusa.com.

Circle #403 on Reader Service Card

▲ MOTU 1224

Mark of the Unicorn's 1224 24-bit digital audio interface for Mac and PC can be purchased as a core system ($1,295) or as an expansion unit ($995) for use with the company's 2408 system or another 1224. The expander consists of the 1224 audio interface; the core system adds MOTU's PCI-324 card for connection to a computer, along with ASIO and Wave drivers. Three 1224s can be connected to a single PCI-324 card.

Bundled with the core system is the *AudioDesk* software for Macintosh, which allows you to record, perform sample-accurate editing and sync, process audio in real time with the included MOTU Audio System plug-ins (you can also use Adobe Premiere plug-ins), and mix entirely within the program using 32-bit floating-point processing.

The rear panel of the interface has eight inputs and eight outputs, all on balanced ¼-inch TRS connectors. In addition, the 1224 has a pair of main outputs, each on a balanced XLR jack. A/D and D/A converters are 24 bit, and you can choose between 16-, 20-, and 24-bit recording at 44.1 or 48 kHz sample rates. Stereo AES/EBU digital input and output is included on a pair of XLR connectors. Word-clock I/O is provided on a pair of BNC connectors. The 1224 can sync to word clock, its digital input, or other clock sources, such as Alesis ADAT Optical, Tascam TDIF, and MOTU's Digital Timepiece. You can record on all 10 input channels simultaneously while playing back 12 channels of audio.

The front panel of the 1224 interface has individual 6-segment level meters for all I/O. It also has a ¼-inch TRS headphone jack with its own volume knob, a power switch, and LEDs that indicate either 44.1 or 48 kHz sample rates. MOTU rates the dynamic range of the analog input and outputs at 116 dB, THD+N at 0.001% (inputs) and 0.0015% (main outputs). Mark of the Unicorn, Inc.; tel. (617) 576-2760; fax (617) 576-3609; e-mail info@motu.com; Web www.motu.com.

Circle #404 on Reader Service Card

▶ WAVES PRO-FX

Waves has added to its line of DSP software with the Pro-FX bundle of effects plug-ins. The bundle is available in TDM format ($700) and in AudioSuite, Premiere, and VST formats ($400 each). The four Pro-FX plug-ins include a pitch shifter and others that emulate various analog devices. The plug-ins all use 56-bit internal processing.

UltraPitch can synthesize from one to six voices and offers formant-based pitch shifting. You can apply delay individually to each voice. You can also adjust pan positions and apply rotary-speaker effects. *SuperTap* is a multitap delay that can apply up to six taps and offers six seconds of delay. It provides EQ filters for each tap, as well as LFO modulation, panning, and more. You can set the tempo and rhythm using an onscreen tap-tempo pad or by entering delay times in milliseconds.

MetaFlanger introduces analog-tape-style flanging effects and comb-filter phase emulations, and you can use it to flange selected frequencies. Finally, *MondoMod* is a tool for creating bizarre effects that combines amplitude and frequency modulation and rotary effects into a single modulation source, with a built-in LFO and the ability to modulate from a sidechain source.

To use the TDM version of Pro-FX, you'll need a Power Mac and Pro Tools TDM hardware, Mac OS 8.1, and 48 MB RAM. All other versions require at least a Power Mac/166, Mac OS 8.0, and 64 MB RAM. Waves; tel. (423) 689-5395; fax (423) 688-4260; e-mail sales-info.us@waves.com; Web www.waves.com.

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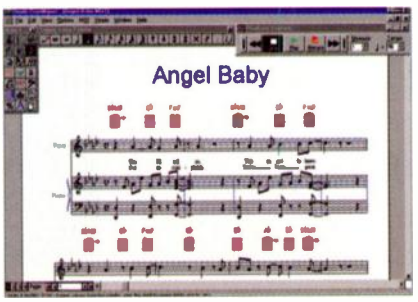
▶ CODA PRINTMUSIC

Musicians looking for an entry-level notation program have a new option in Coda's *PrintMusic* (Win; $69.95), the newly released "lite" version of the company's *Finale* software. *PrintMusic* provides the tools to input basic scores with a computer keyboard and mouse or with any MIDI controller. The program can import Standard MIDI Files, as well. You can play files back using your sound card's General MIDI sounds or through your MIDI instrument.

PrintMusic places notes on your score instantly as you enter them. The features it offers include guitar tablature, lyric entry, and transposition. The program comes with more than 100 pieces of music that you can use to practice arranging and manipulating formats. All the tools in this program are equivalent to those found in Coda's *Allegro* and *Finale* software, so the skills you learn and the music you create will easily translate to these more sophisticated programs.



Bundled with the program is the *SmartMusic Studio* package, which features vocal and instrumental practice tools, such as *Vivace Intelligent Accompaniment*, a program that provides interactive instrumental backup.

You can run *PrintMusic* on Windows 95, 98, or NT, and it requires at least 16 MB RAM. Coda Music Technology; tel. (800) 843-2066 or (612) 937-9611; fax (612) 937-9760; e-mail finalesales@codamusic.com; Web www.codamusic.com.

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SOUND ADVICE ▲▲▲▲

Q UP ARTS

For those who seek true-to-life piano sounds, Q Up Arts says, "The quest is over." The company's new CD-ROM *The Holy Grail* ($499) is packed with piano samples in Akai, E-mu, or Roland formats. (The samples have also been licensed to E-mu, which will make them available on a 16 MB card for the Proteus 2000.) Q Up Arts has spent two years, in collaboration with Alpha-Omega, developing a sampling method that is called Virtual Model Sampling. This technique was applied on *The Holy Grail* to provide more realistic piano

sounds that capture a high degree of the natural sympathetic resonance and full sustain of each note.

The samples on *The Holy Grail* were recorded at 24-bit resolution. The disc gives you control over many attributes of the instrument's sound, including pedal and hammer action, sympathetic string vibration, and note attack and release sounds. For added flexibility, the entire range of the keyboard was recorded with two different mic placement setups and in velocities ranging from mezzo piano to fortissimo. Multiple versions in varying memory sizes are provided to accommodate users' RAM availability.

Sonic components of the whole instrument—such as pedal noise, note attack, and harp resonance—are assigned to different channel outputs in order to make full use of surround-sound capability and to create a complete, three-dimensional picture of a grand piano. Q Up Arts; tel. (800) 454-4563 or (530) 477-8128; fax (530) 477-5935; e-mail sales@quparts.com; Web www.quparts.com.

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HRUSKA AUDIO PRODUCTIONS

Sonic Implants ($9.95 to $34.95), from Hruska Audio Productions, are downloadable sound libraries and Standard MIDI Files for PC users. You can purchase individual sounds or complete libraries of these SoundFont-format files and use them with Creative Labs' Sound Blaster sound cards, Steinberg's *Cubase*, E-mu's Audio Production Studio, and Windows-based software synthesizers that read SoundFonts, such as Seer Systems' *Reality* and *SurReal*.

Industrial Dance PC and *Retro Synth PC* are GM wavetable sets with industrial- and analog-synth-style instrument substitutions. *Session Drums* and *Brush Drumkits* are stereo multisamples of an entire drum kit recorded with sticks and brushes, respectively. The *Brush Drumkits* group features snare hits, swirls, taps, and strokes, as well as brushed toms and cymbals. The drum libraries are laid out in GM keymaps, as is *Processed Kicks and Snares*, a collection of 20 kicks and 20 snares. Similar libraries contain stereo and mono multisamples of grand pianos, with different libraries optimized for E-mu APS systems and Creative Labs sound cards.

Other groups include four sets of electronica material; acoustic and electric guitar, bass, and brass sets; and *Songwriter 1 Classics*, which combines various instrument samples and MIDI files containing grooves and fills. Hruska Audio Productions; tel. (781) 641-0063; fax (781) 641-0448; e-mail jhruska@hruskaudio.com; Web www.hruskaudio.com.

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ANALOGUESAMPLES.COM

Do you wish that you had never gotten rid of that old analog synth? Or have you recently caught on to vintage analog sounds and want to use them in your music? You may have been frustrated to discover that those relics now cost an arm and a leg—if you're lucky enough even to find one. Fortunately, you can get some of those sounds for free by surfing over to Analoguesamples.com.

The founders of this Web site have graciously set up a library offering WAV files of sounds from classic synths and drum machines, contributed by musicians from around the world. There are currently more than 1,000 files available, with the number growing steadily. All downloads are free of charge and require no licensing for further use.

You can search for sounds by category (bass, pads, chords, lead, and so on), type of synth, contributing artist, memory size, or upload date. The sounds are grouped into "packs" containing anywhere from 1 to 40 samples in the same category from the same synth. You can't audition the sounds on the Web site, but because you can download them for free, you can't go wrong. Analoguesamples.com; www.analoguesamples.com.

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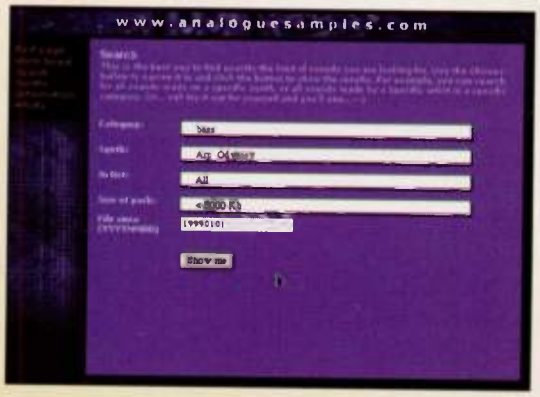

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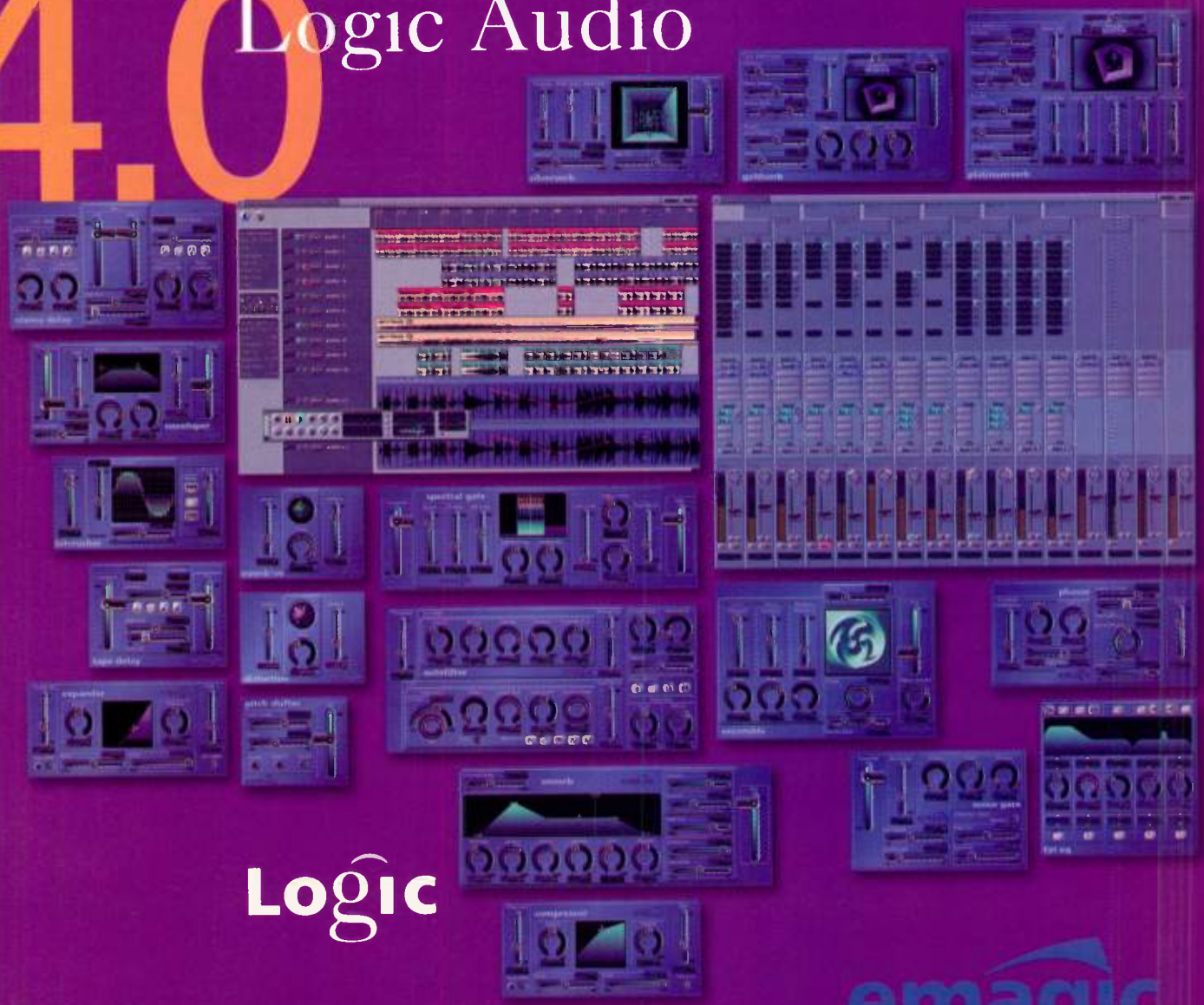

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I WANT MY MP3 ▲▲▲▲

MAYAH

Mayah has announced *EditPro* ($600), a Windows-based audio-file editor that works directly on compressed MPEG Layer 2 and MPEG Layer 3 files. According to Mayah, this method eliminates the degradation created each time an audio file is recompressed.

In addition to editing MPEG 2 and MPEG 3 files, *EditPro* supports the broadcast version of the WAV file format and raw MPEG files. The program's editing functions include cut, copy, paste, crossfading, and others.

The program can be purchased as part of the Compact Studio bundle ($1,625), which also includes *SendIt* ($1,150 when purchased separately), a new software codec from Musicam designed for compressing and expanding MP3 files. *SendIt* can send and receive 20 kHz stereo audio over an ISDN line when connected to most hardware codecs, or to a *SendIt*-equipped PC. The program can also send and receive audio in real time over an analog phone line when connected to other *SendIt*-equipped PCs. In addition to sending and receiving audio in real time, *SendIt* can encode audio in MPEG 2, MPEG 3, broadcast WAV, and Digigram Musifile formats.

EditPro requires a Pentium/333 with a minimum of 24 MB of RAM. Musicam USA (distributor); tel. (732) 739-5600; fax (732) 739-1818; e-mail info@musicamusa.com; Web www.musicamusa.com.

Circle #410 on Reader Service Card

XING TECHNOLOGY

Mixman Technologies is distributing the new *AudioCatalyst* MP3-encoding software (Mac/Win; $49.95) from Xing Technology. Combining Xing's *AudioGrabber* and *MP3 Encoder*, *AudioCatalyst* can take stereo or mono audio from a CD and transform it into an MP3 file all in a single step, in less than real time.

AudioCatalyst provides an extended frequency response (up to 20 kHz), and the program gives you the option to add silence at the beginning or end of a file. In the Windows version, you can select start and stop times if you want to encode just a segment of your audio, and that version also has a volume-normalizing function. *AudioCatalyst* includes the *XingMP3 Player*, which offers playlist management in a database where you can fill in artist, album, and track information.

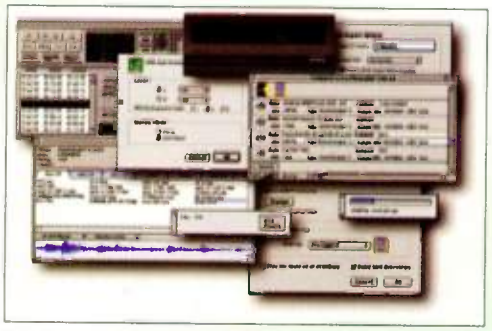
In addition to encoding CD Red Book audio to MP3, *AudioCatalyst* can save Red Book audio in WAV (Windows) or AIFF (Mac) format, or encode WAV files as MP3. It can also encode CD audio to both WAV and MP3 simultaneously. The resulting MP3 files comply with the ISO standard, so they will play back on any MP3 player. *AudioCatalyst*'s compression scheme incorporates Xing's Variable Bit Rate encoding, rather than a constant bit-rate method, to help reduce file sizes. The program supports data transfer rates from 32 to 320 kbps.

Minimum system requirements for the PC version are a Pentium CPU with at least 16 MB of RAM, running Windows 95/98/NT 4.0, with any standard sound card. For Macintosh, *AudioCatalyst* requires a PowerPC computer with Mac OS 7.5 or higher, 24 MB of RAM, and the Apple-CD or AppleCD/DVD driver. Mixman Technologies, Inc.; tel. (415) 403-1382; fax (415) 403-1388; Web www.mixman.com or www.xingtech.com.

Circle #411 on Reader Service Card

GALLERY SOFTWARE

Gallery Software's mTools ($849), a stand-alone suite of programs, is geared toward use with Digidesign's *Pro Tools* audio editor, but it can be used with other digital audio sequencers, as well. The first component, *dCode*, takes MPEG files and converts them to SDII files, automatically splitting stereo interleaved files and building SDII overviews. You can drag and drop MPEG files on the *dCode* icon, and when processing is complete, the program drops the new files in *Pro Tools'* Region bin.



Using the mCode component, you can drag and drop files for encoding into MPEG format. *SampleSearch* 2.2 aids in searching for files, letting you search for key words, file types, creation dates, and other categories. Finally, you can use *CDStudio* 2.5 and *SFX Database Builder* to develop and manage MPEG file libraries, catalog your files, and reference a file's sample rate and duration.

SampleSearch and *dCode* are both controllable by using AppleScripts, and included with the mTools package is a sample *FileMaker Pro* database with examples of script buttons to help you get started. Gallery Software; tel. 44-171-431-6260; fax 44-171-435-8134; e-mail information@gallery.co.uk; Web www.gallery.co.uk.

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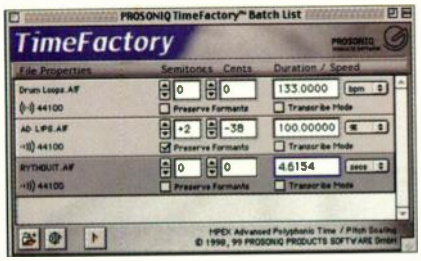


PROSONIQ TIMEFACTORY

Prosoniq's *TimeFactory* (Mac/Win; $590) is a stand-alone application that uses the company's Minimum Perceived Loss Time Compression/Expansion. This proprietary algorithm provides high-quality time stretching and pitch manipulation of digital audio tracks without introducing phasing or timing problems. The program supports 8-, 16-, and 24-bit files, as well as resolutions of up to 96 kHz. You can process WAV, Sound Designer II (Mac version only), and AIFF mono and stereo files.

According to Prosoniq, *TimeFactory* can stretch a track to at least 133 percent of its original length or, alternately, shift its pitch up or down by as much as five semitones, without loss of audio quality or introduction of unwanted artifacts. You can apply a time adjustment by setting parameters in various ways: you can select the new length in seconds, in bpm, or as a percentage. To apply a pitch shift, you choose the number of semitones and cents to shift, and the new length of the track, within the program's main window.

You can also choose whether to preserve the formant of the original signal; *TimeFactory*'s new proprietary timbreWizard algorithm is used in the process for more natural-sounding pitch shifts. The program can process files individually or in batches.

For the Macintosh, *TimeFactory* requires a 603e Power PC processor running Mac OS 8.0 or higher, and 16 MB of RAM. The Mac version ships with Prosoniq's *SonicWorx Essential* stereo audio editor. PC users need at least a Pentium II/266 running Windows 95, 32 MB of free RAM, and DirectX 5.0. Steinberg North America (distributor); tel. (818) 678-5100; fax (818) 678-5199; e-mail info@steinberg-na.com; Web www.us.steinberg.net.

Circle #413 on Reader Service Card

SEER SYSTEMS SURREAL

Seer Systems' *SurReal* (Win; $129) is a software-based sampler and synth capable of FM, physical modeling, analog-style, and modal synthesis (a form of physical modeling useful in creating bell sounds). The program is capable of playing up to 16 sounds simultaneously, with maximum polyphony of 64 notes, depending on your PC's processing power. *SurReal* can also load SoundFonts.

From *SurReal*'s main graphic interface, you can assign up to 16 tracks and individually adjust each track's volume and pan position. Up to two real-time effects can be used per track; effects include reverb, chorus, delay, flange, and others. There is a knob for control over each effect's wet/dry mix. Each channel also features a Vari function that makes intelligent parameter changes based on the chosen sound. There are also solo and mute buttons for each track. A stereo master-control window lets you adjust overall volume and panning, and it gives you the ability to apply up to two real-time master effects.

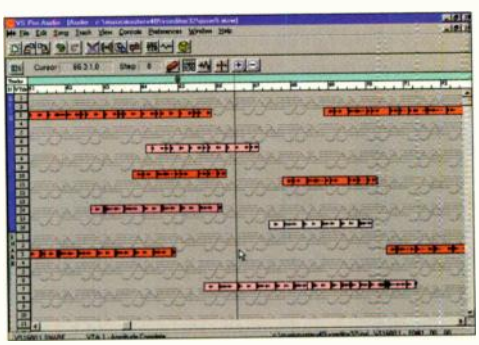
The program ships with a full GM sound set, eight GS-compatible drum kits, and many other sounds. You can play MIDI sequences from within *SurReal* and save the output as audio files. You can also create SeerMusic files, which are low-bandwidth sound files meant for the Web. To play these, you'll need to download the free SeerMusic player from Seer System's Web site. *SurReal* requires at least a Pentium/133 running Windows 95 or 98, and 32 MB of free RAM. Seer Systems; tel. (888) 232-7337 or (650) 947-1915; fax (650) 947-1925; e-mail sales@seersystems.com; Web www.seersystems.com.

Circle #414 on Reader Service Card

DATASONICS VS PRO

Datasonics' VS Pro software plug-in suites allow you to track, mix, edit, sequence MIDI tracks, and apply effects on Roland VS-880, VS-880EX, or VS-1680 digital recorders using a standard MIDI interface and a Windows PC. The program uses Control Change and SysEx messages to exchange information, so all of the actual editing is done within the VS recorder.

VS Pro Score ($399) is a host program with several plug-ins that control different aspects of the Roland recorder's functions. You can use *Global Editor* and *Audio Editor* to perform standard cut, copy, and paste functions for MIDI and audio tracks, as well as audio waveform editing. Other plug-ins let you control the VS recorder's channel EQ, effects parameters, and other audio functions. *Notation Editor* can display your MIDI tracks in standard notation, percussion staves, guitar chord charts, and more, and you can print your score from within the program. Other dedicated plug-ins let you control synchronization, MMC transport, and general VS recorder settings.



VS Pro Studio ($299) includes everything in *Pro Score* except the *Notation Editor*, while *VS Pro Lite* ($129) includes just the basic sequencer. The other plug-ins can be purchased individually as *FX Tools* ($69), *Audio Tools* ($99), *Mix Tools* ($99), or together as the *ToolBox* ($199); these can be used with third-party sequencers. The programs require at least a Pentium/90 running Windows 95, 98, or NT with 16 MB of free RAM. Datasonics; tel. 61-2-4758-7755; fax 61-2-9225-9010; e-mail sales@datasonics.com.au; Web www.datasonics.com.au.

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ON THE HORIZON ▲▲▲▲

ALESIS

The MasterLink ML-9600 ($1,699) from Alesis is a high-resolution, stereo hard-disk recorder and CD-R burner created for use as a stand-alone mixdown and mastering solution for the personal studio. The MasterLink combines a 3.2 GB IDE hard drive, Matsushita 4x CD recorder, 24-bit digital converters, editor, and internal DSP effects into a single, 2U rack-mountable unit.

The ML-9600 can create CDs in two formats: standard audio (Red Book) and high-resolution CD24. The CD24 is a proprietary Alesis format that writes AIFF files in any combination of sample and bit rate (up to 24 bit/96 kHz) to regular CD-R discs. These files can be read by the MasterLink or any computer with a standard ISO 9660 CD-ROM drive.

The MasterLink hard disk can handle 95 minutes of 24-bit/96 kHz stereo audio, and 310 minutes of stereo 16-bit/44:1 kHz audio. When you're ready to burn a disc, choose your format and hit the Create CD button. The CD24 format yields about 20 minutes of 24-bit/96 kHz audio per disc.

MasterLink stores 16 playlists with up to 99 tracks in each. You have your choice of destructive or nondestructive editing, with three crossfade shapes to choose from, and the ability to name tracks. Although, you can't print playlists directly from MasterLink, playlists are automatically written to CD24 discs.

The DSP section includes useful effects such as a compressor, a 3-band parametric equalizer, a normalizer, and a peak limiter. Other DSP effects will be added in future software upgrades or as proprietary, third-party plug-ins.

The unit has 24-bit digital converters with 128x oversampling. Analog connections include +4 dBu balanced XLR and -10 dBV unbalanced RCA jacks. The digital I/O uses AES/EBU and S/PDIF-style coaxial connectors. There is no SCSI connector or word-clock I/O on the MasterLink.

The dynamic range of the ML-9600 is 113 dB. At sample rates of 96 and 88.2 kHz, it has a frequency range of 20 Hz to 40 kHz. At 44.1 and 48 kHz the frequency response is 20 Hz to 20 kHz. The unit can record audio in any combination of the four sample rates (44.1, 48, 88.2, and 96 kHz) and three word lengths (16, 20, and 24 bit). The ML-9600 also converts formats, so a playlist can contain files in different bit rate/word length combinations, and the unit will convert each file to the format chosen for the CD-R. Alesis plans to have MasterLink available in the fourth quarter of 1999. Alesis Corporation; tel. (800) 525-3747 or (310) 255-3400; fax (310) 255-3401; e-mail alecorp@alesis1.usa.com; Web www.alesis.com.

—Gino Robair

Circle #416 on Reader Service Card

SONIC TIMEWORKS

Sonic Timeworks has released four new DirectX plug-ins: a reverb, a digital delay, an 8-band equalizer, and a phaser. All feature hardware-style interfaces and real-time auditioning, with 64-bit internal processing and support for sample rates up to 192 kHz.

Reverb Model 4080L ($179), which is also available for CreamWare Pulsar/SCOPE, provides stereo, mono, and dual-mono processing. It comes with numerous presets, and you can customize as many additional presets as your system will allow, with adjustable settings for decay, predelay, damping, color, and room size.

The *Delay 6022* plug-in ($79) gives you virtual sliders to control delay time and spin (feedback) time, as well as output level. These parameters can be set individually for the left and right channels, or you can gang the two channels to lock their settings together. A wet/dry mix slider, a gain slider, and left and right tone control knobs are also provided. The VU-style meter can be set to display input, output, or delay level and has a clip indicator.

Timeworks Mastering EQ ($249) provides six bands of paragraphic EQ with sweepable Q, bandwidth, and adjustable cut/boost (6, 12, 18, or 24 dB) on each band. Parameters can be set using color-coded sliders or numeric entry. There are high and low shelving filters, also with adjustable cut/boost amount and frequency.

Lastly, *Phazer Model 88* ($79) features sliders for controlling depth, feedback, rate, and wet and dry output levels. Phasing frequency is also selectable, and a gain slider is accompanied by a clip indicator. *Phazer Model 88* can operate in mono, stereo, or dual mono.

These plug-ins can also be purchased together as a bundle for $479. They require a Pentium PC, Windows 95/98/NT, 16 MB of RAM, and any DirectX-compatible host program. VST versions of these plug-ins are being developed. Sonic Timeworks; tel. (314) 352-8668; fax (314) 832-5308; e-mail sales@sonictimeworks.com; Web www.sonictimeworks.com. ●

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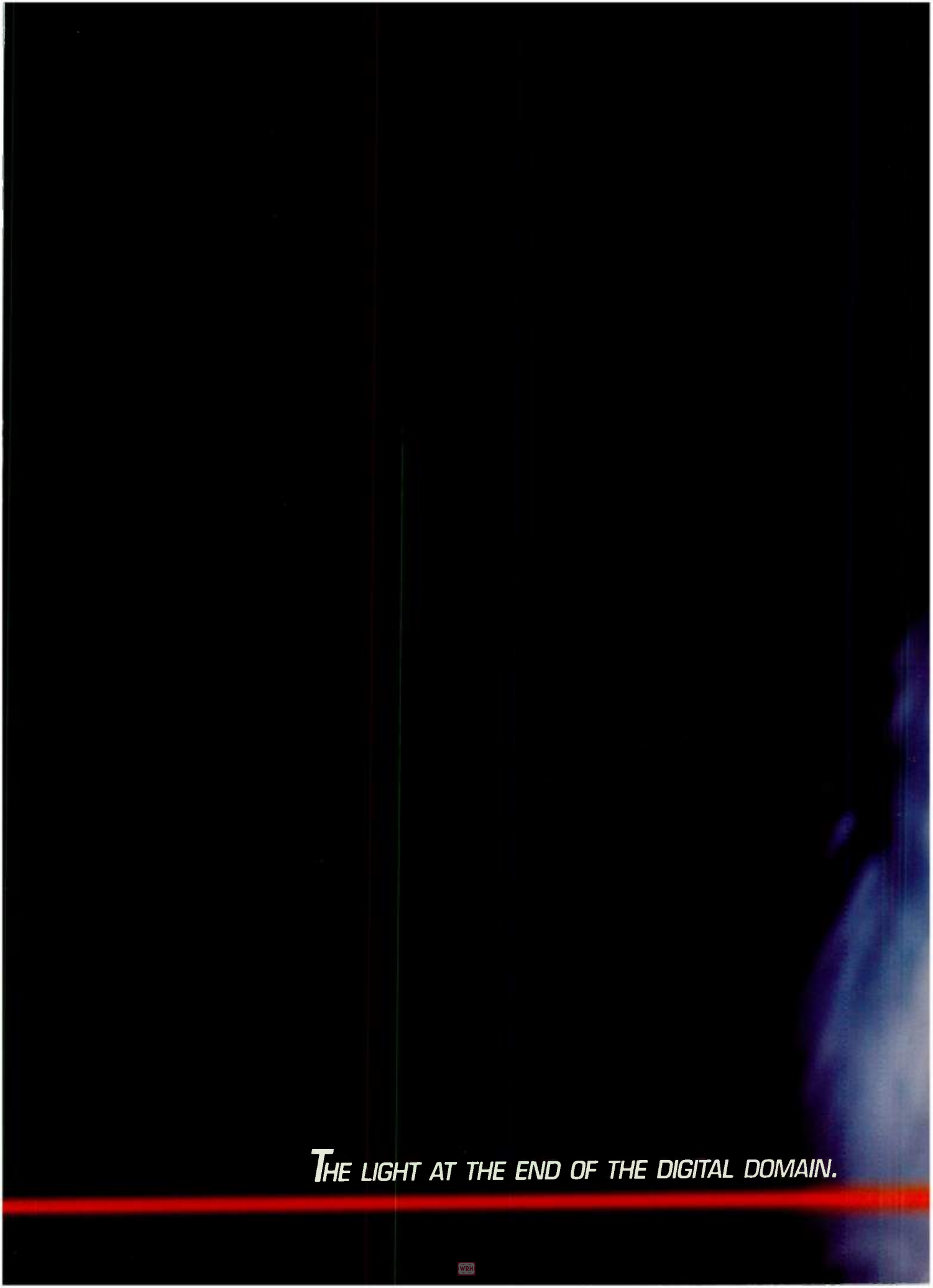
THE LIGHT AT THE END OF THE DIGITAL DOMAIN.

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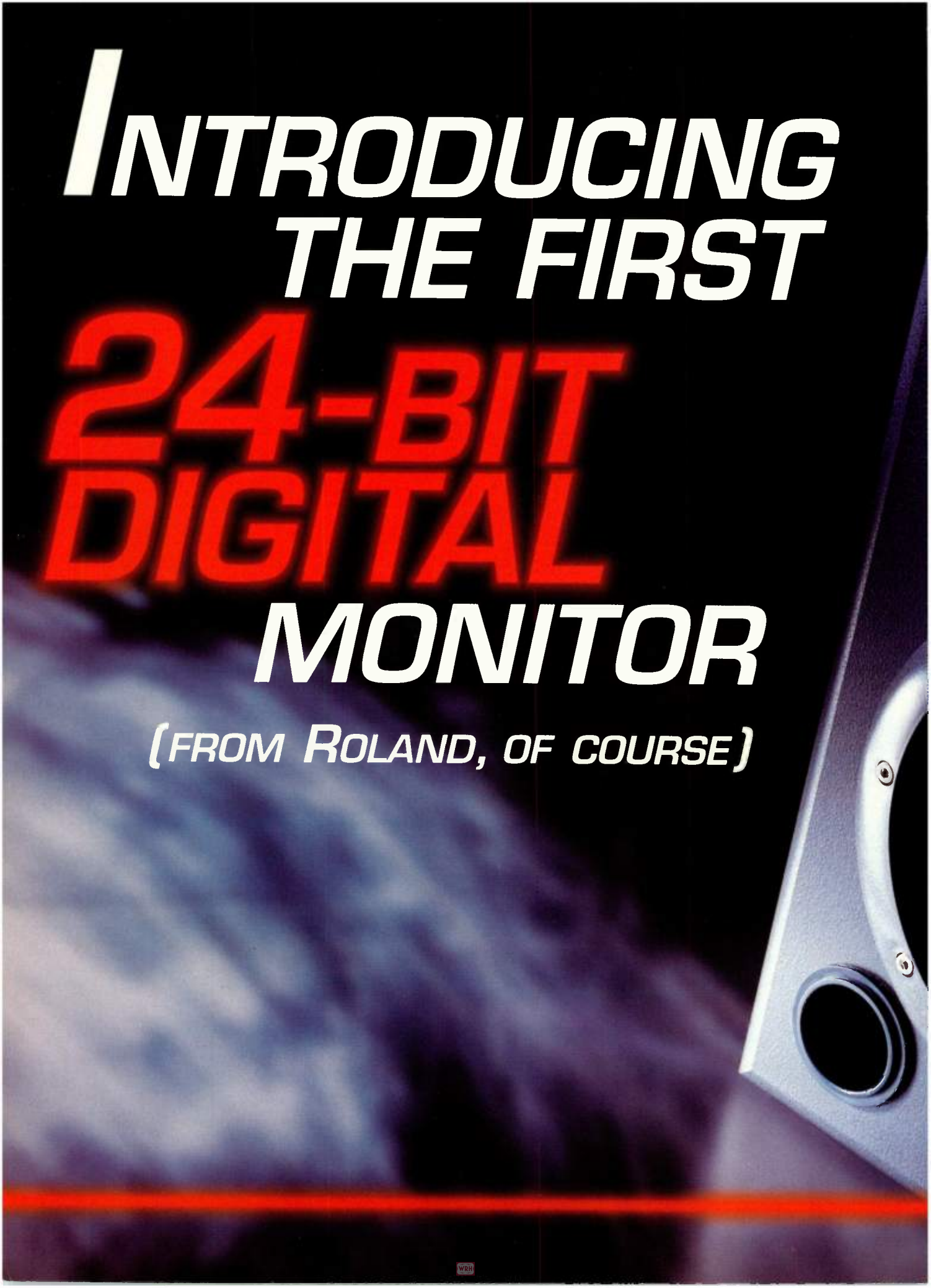
(FROM ROLAND, OF COURSE)



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Continuous noise consists of any sounds that continuously occur in the audio signal but are unrelated to the source material. These noises include anything from tape hiss and static to the sound of an air conditioner or computer fan in the background. Unless you have a period of relative silence in your source material, you can't really see continuous noise in a waveform display, because it doesn't stand out the way that impulsive noises do.

Harmonic noise is a type of continuous noise that contains a specific fundamental frequency and sometimes a number of odd or even harmonics. Harmonic noises are usually caused by ground loops, light dimmers, fluorescent lighting, or AC power supplies. They are more commonly known as *hum* or *buzz* and typically occur at 60 Hz (the frequency of AC in the United States) or 50 Hz (the frequency of AC in Europe). You can view harmonic noise as a pronounced "bump" at the appropriate frequency in a normal audio waveform.

Because these noise types have specific and identifiable characteristics, it's possible to reduce their presence (and often eliminate them completely) in an audio signal without degrading the source material. In certain cases, you may need only an audio-editing program with common functions such as cutting, pasting, crossfading, and EQ. But most of the time, you'll need one or more tools that are specifically designed to deal with these sonic disturbances.

SHOW ME THE SOFTWARE

A number of applications now offer powerful tools for eliminating noise from audio recordings. Most of the current programs are available for the PC, but many are also available for the Mac. Some products come as plug-ins for use with audio-editing software, others are stand-alone applications created specifically for audio restoration, and still others are available in both forms.

FIG. 2: Arboretum's inexpensive *Ray Gun* offers several types of noise reduction in a single, easy-to-use plug-in.

If you already own a high-end audio-editing program, such as Sonic Foundry *Sound Forge*, Steinberg *WaveLab*, or BIAS *Peak*, then it makes good sense to think about purchasing a noise-reduction plug-in. Digidesign Pro Tools users might consider *Digidesign Intelligent Noise Reduction (DINR)*. It provides functions for continuous-noise reduction and harmonic-noise cancellation. Unfortunately, it's available only as a TDM plug-in. (An AudioSuite version is also available, but it lacks the Hum Removal function.) And at $995, *DINR* is also somewhat expensive.

Another high-end option for the Macintosh is Arboretum's *Ionizer* ($499), a powerful audio-processing and spectral-reshaping tool that comes in stand-alone form as well as in Premiere and AudioSuite plug-in versions. (A MOTU Audio System version is due out soon.) In addition to its compression, expansion, and limiting capabilities, *Ionizer* offers broadband noise reduction and a 512-band gated equalizer.

On Windows systems, DirectX-based products are often the best choice. DirectX plug-ins work with most Windows-compatible audio software, and in many cases, they are comparatively easy on the wallet. DirectX offerings include Steinberg's *DeClicker* and *DeNoiser*, Arboretum's *Ray Gun*, and Sonic Foundry's *Noise Reduction* package. Arboretum also offers *Restoration-NR* ($299), a DirectX plug-in that specializes in high-resolution hiss removal. *DeClicker* and *DeNoiser* are available for the Mac in Steinberg's VST plug-in format (*DeClicker* is also available in TDM format); *Ray Gun* is available in VST, Premiere, and AudioSuite formats as well as in stand-alone versions for Mac and Windows.

At $399 each, *DeClicker* and *DeNoiser* are the most expensive plug-ins in the lot, and they're only available separately. (For full reviews, see the April 1998 issue of **EM**.) They stand out from other noise-reduction products because they're so easy to use. With *DeClicker*, you just choose a Mode, set the Threshold, and select the Click Reduction quality. *DeNoiser* is similarly streamlined, with settings only for Level, Reduction, and Ambience (see **Fig. 1**). In addition, *DeNoiser*, unlike most other noise-reduction software, doesn't rely on a noise print (a section of the audio containing only noise without the source material). *DeNoiser* analyzes the audio file and makes an educated guess as to which part of the signal is noise. Even with their simplified controls, the *DeClicker* and *DeNoiser* processing algorithms are quite complex, and their end results are very good.

Ray Gun ($99) is the least expensive plug-in of the group. It offers broadband noise reduction along with pop, rumble, and hum filters in a single package (see **Fig. 2**). In other respects, it's much like the Steinberg plug-ins, providing simplified controls and dynamic detection algorithms. For noise reduction, you can set Threshold and

FIG. 1: Steinberg's *DeNoiser* plug-in provides sophisticated noise-reduction capabilities with a simple user interface.

For manufacturer information, please see Contact Sheet, p. 163.

Attenuation levels. The pop filter has a single Sensitivity setting, and the rumble and hum filters can simply be turned on or off. There's no way to adjust the frequency or bandwidth of the filters.

Sonic Foundry's *Noise Reduction* plug-in package is a good deal. The $349 package includes three plug-ins: *Noise Reduction, Click Removal,* and *Vinyl Restoration.* These plug-ins allow you to attack all three categories of noise, and they provide plenty of control over how the signal is processed. For impulsive noises, the *Click Removal* plug-in lets you find each click individually and eliminate it manually or automatically by replacing or interpolating the signal (more on these options later). For continuous and harmonic noises, the *Noise Reduction* plug-in has adjustable attack and release speeds, FFT size (up to 16,384 points), and high shelf gain. It also allows you to fine-tune the detected noise envelope and accentuate or attenuate individual frequency points. The *Vinyl Restoration* plug-in offers special features that are tailored toward cleaning up old vinyl recordings (although you can use the other two plug-ins to do the same thing).

If you would rather purchase a stand-alone noise-reduction program, your choices include *Clean* from Steinberg, *CD-Recorder* and *DartPro 98* from Dartech, and *DC-ART 32* from Tracer Technologies. These products are available only for the PC. (As mentioned earlier, Arboretum's *Ray Gun* is also available for the Mac and PC in a stand-alone version.) *Clean* and *CD-Recorder* are CD-recording applications with noise-reduction capabilities. *Clean* ($129) provides simplified click-, crackle-, and noise-removal functions—each with a single Intensity control. *Clean* also includes a copy of Steinberg's *WaveLab Lite* audio editor. *CD-Recorder* ($49.95) comes with a package of five DirectX plug-ins that include simple click- and hiss-removal plug-ins. They provide a bit more control than *Clean,* although the program lacks waveform editing functions. Nonetheless, at less than $50, it may be worth buying for the plug-ins alone.

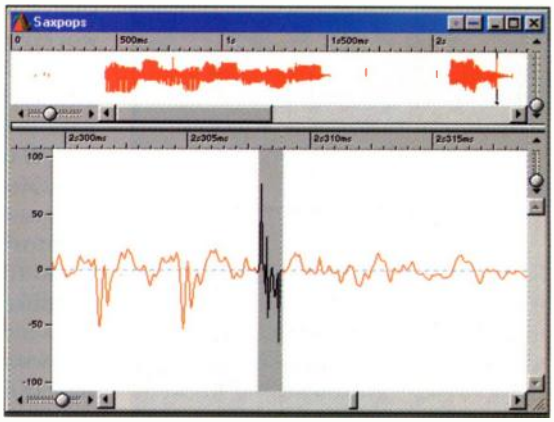

FIG. 4: Impulsive noises, often called clicks or pops, are fast transients that look like spikes in the normal curve of a waveform.

DartPro 98 and *DC-ART 32* are full-featured noise-reduction applications. With a price tag of $399.95, *DartPro 98* is among the more expensive restoration programs but is also one of the most powerful. It provides many of the same functions as Dartech's earlier *DartPro 32 (DartPro 98* is offered as an upgrade to *DartPro 32),* although it also has many additional features. Its processing tools are likewise more advanced. *DartPro 98*'s audio-restoration features include DeClick, DeNoise, DeHiss, and ReTouch, (for filling gaps and dropouts). The FilterBuilder lets you create your own sequence of restoration filters and save them as presets.

DartPro 98 includes a hum-removal tool for dealing with continuous harmonic noises, and Smart Mute and Smart Cut tools thst let you quickly and easily remove small disturbances. The application's noise-removal tool lets you reshape the noise-print curve. *DartPro 98* supports DirectX plug-ins, and its real-time processing allows you to plug an audio source into the computer and store the processed sound to an external medium without engaging the hard disk. You can also turn the onscreen knobs and make adjustments while listening to the results. The program's audio-editing and processing tools include Fade, Equalizer, Gain, Resample, and 2D/3D Spectral Analysis. Moreover, *DartPro 98* includes a copy of *CD-Recorder,* so you can burn a CD after restoring your files.

DC-ART 32 costs half as much as *DartPro 98,* but this $199 program provides much of the same functionality (see **Fig. 3**). In fact, it's even *more* powerful in a few areas. In addition to the usual impulsive, continuous, and harmonic noise-reduction tools, *DC-ART 32* offers sophisticated paragraphic equalization, dynamics processing (expander/gate, de-esser), and reverb, as well as tube-amplifier-emulation effects. Too bad it doesn't support DirectX. The *DC-ART* documentation, however, is outstanding, describing every feature of the program, as well as explaining how and why each function works.

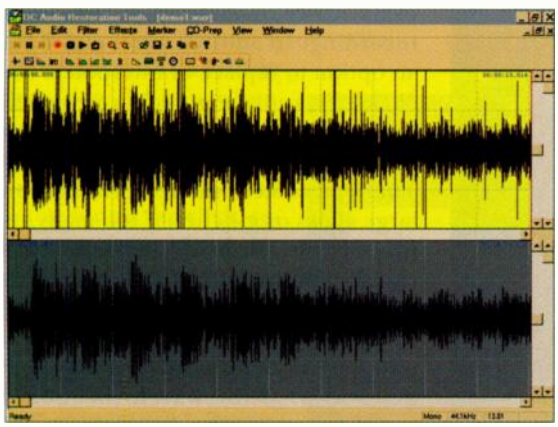

FIG. 3: Tracer's *DC-ART 32* offers several powerful noise-reduction capabilities in a moderately priced program. In this example, a waveform is shown before and after the automatic click-removal process is applied.

TAKE OUT THE TRASH

Unfortunately, there are no hard-and-fast rules when it comes to removing noise from your files, because every noisy recording is different. There are, however, techniques that you can use (in a particular order) to give you the best chance at successfully cleaning up your audio.

The first thing to consider is which sampling rate to use. You may think that the higher the sampling rate, the better; many people believe that audio files should always be recorded using at least the standard CD-quality sampling rate of 44.1 kHz. After all, the more samples taken, the better the representation of your audio signal. Other people, however, argue that because most old tape and vinyl recordings don't have a frequency response higher than 10 kHz, a 22.05 kHz sampling rate is fine. In some cases, it may even

the noise print, the better. A good size is from 1 to 2 seconds, but if necessary, you might get by with 400 samples or less. (Many programs use a preset window of 1,024 samples.)

After you've captured some noise, you'll see a graph showing the amplitude versus the frequency of the selected noise. You'll also see a noise threshold contour (see **Fig. 6**). This contour, or envelope, controls the amplitude threshold below which any signal is attenuated and above which any signal is passed through without processing. Ideally, you should set the threshold slightly above the noise print; many programs allow you to adjust the threshold level. Be careful not to set it too high, or you'll start to cut out some of the source material.

One cool feature that some programs offer is called "Keep residue" or "Keep residual output." Activating this option lets you listen to the part of the signal that's being removed (the noise), rather than the part of the signal that you're trying to save. Listening to the residual output as you adjust the threshold lets you hear whether you're taking part of the source material with it.

After you have set the threshold, you still must adjust several other parameters to get the right amount of noise reduction without adding artifacts to the source material. The attack-time setting tells the noise-reduction algorithm how fast to react to the audio signal. If you set this too high, you might introduce artifacts into the signal. A good initial setting is about 25 milliseconds. The release-time parameter indicates how quickly the signal will be tapered off. A good initial setting for this is about 50 milliseconds. If you hear the ends of long-decaying sounds being trimmed off, increase the value.

The attenuation setting is the most important control. It tells the noise-reduction algorithm how much to reduce the noise level. You must be careful when using this adjustment because too little attenuation leaves noise in the recording, while too much introduces artifacts into the signal. A good initial setting is around 10 to 20 dB. If your software allows it, you can have the recording play while you adjust the attenuation. That helps attain the proper setting more quickly.

Making two passes with lower attenuation settings may be better than making a single pass at a higher setting. In other words, if you have a particularly noisy file, grab a noise print and process the file with an attenuation of about 10 to 20 dB. Then grab a new noise print and process the file again with the same attenuation setting. This doubles the amount of noise that is removed but is less likely to introduce unwanted artifacts into the signal. On the other hand, you also run the risk of introducing new calculation errors into the signal with multiple passes. If your noise-reduction system introduces any aliasing (which might not be readily apparent at first), you might wind up doubling the aliasing by doing two passes. The best policy, therefore, is to experiment to see which approach delivers the best results with your setup.

HARMONIC NOISES

Because harmonic noises are a form of continuous noise, some or all of them may have been removed during the previous denoising process. If they weren't removed, however, a bit of judicious equalization should do the job.

Some software applications provide hum removal or harmonic rejection. These tools are essentially a string of notch filters grouped together into one big multifrequency filter, often called a "comb" filter. If your software has this tool, give it a try. Otherwise, you can use a simple notch filter (which most audio-editing programs offer) to remove hum or buzz. The process can get a bit tedious, however, because even though most hum or buzz noises sound the strongest at one fundamental frequency, the fundamental is often accompanied by a number of odd or even harmonics. With a specialized hum-removal tool, you just specify the fundamental frequency (in most cases 60 Hz), and the software automatically filters out the odd or even harmonics (up to 500 in some programs). If you use a simple notch filter instead, attenuating enough of the harmonics to adequately eliminate the buzz may take you a while.

Nonetheless, if you're stuck having to remove harmonic noises manually, you may be able to get reasonably good results by notching out the fundamental frequency and a few of the harmonics. In that case, you can use a tool called a paragraphic equalizer, which enables you to set up your own little comb filter; the number of bands

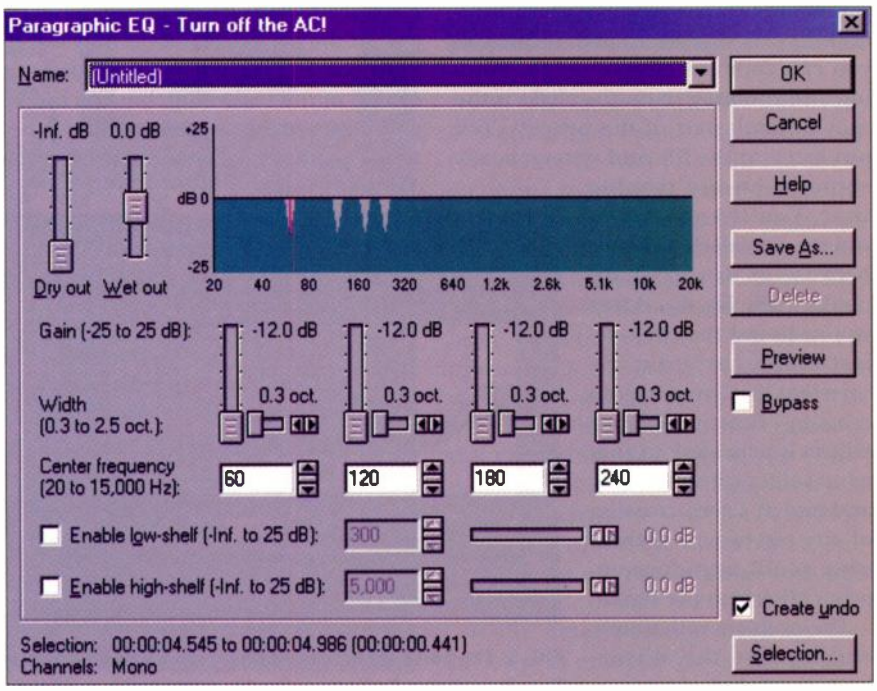

FIG. 7: A paragraphic equalizer lets you set up your own comb filter; the number of bands depends on your software.

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SPECIFICATIONS: ***Amplifier power:*** 150W rms/ch, 300W (100ms peak) ***Peak acoustic output:*** 116dB SPL (100 ms pink noise @ 1M). ***Residual hum/noise:*** ≤10 dB SPL (A-weighted @ 1M). ***THD @ 90 dB SPL:*** <0.8% (100Hz - 10kHz @1M) ***Response: ± 2 dB (1/3 oct. swept noise):*** 57Hz - 20kHz @ 1M, 55Hz - 20kHz @ 2M. ***-6dB LF cutoff:*** 50Hz. ***Control Amp Dimensions/Wgt:*** 3.5"h x 19"w x 10.75"d (2u), 29lbs.

[*Manufacturers suggested retail thinking charge: $1,250.00]

SYSTEM: ***Type:*** Modular active, near/mid-field monitor. ***Woofer:*** 6.5" treated paper. ***Tweeter:*** Soft dome. ***Magnetic Shielding:*** Full. ***Monitor Dimensions/Wgt:*** 11.9"h x 7.5"w x 10.7" d, 14 lbs. ***Monitor Enclosure Materials:*** 3/4" mdf w/HP laminate). ***FEATURES/CONTROLS: Connectors: Input:*** XLR, TRS ***Output:*** XLR. ***Controls: Input sensitivity:*** -10, -3, +4, +11, ∞ dBu. ***Listening proximity:*** 2-position (near/mid) ***Boundary proximity:*** 2-position (0, 1).

A-10 The control amplifier standing on its left ear not only powers the rather handsome monitor pictured in center page, it represents a fairly radical departure in conventional thinking. By bringing the amp outboard, the controls are now where you can get to them, making a snap out of adapting monitor response to listening room acoustics. It also keeps the monitor enclosures themselves from approaching the size of two parked Buicks. Their unusual cabinet shape also improves midrange clarity, stereo imaging and localization. And because its an active monitor, the system response is surprisingly accurate, detailed and has tremendous bass extension. Best of all, considering they are free with the purchase of some pretty solid thinking, the price couldn't be more right.

NHT PRO

www.nhtpro.com

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depends on your software. For example, if I were using *Sound Forge* to eliminate some ground-loop hum, I would set up its Paragraphic EQ tool with four notch filters set to 60, 120, 180, and 240 Hz (see **Fig. 7**). Then I would experiment with the gain and bandwidth settings to find the right combination for removing the hum while affecting the source material as little as possible.

Generally, it's best to cut the gain as much as you can. Start with -6 dB and go from there. Useful settings for bandwidth are between 0.1 and 0.5 octaves (although *Sound Forge* doesn't go below 0.3). You start cutting into the source material with anything higher than 0.5. If some hum is still left after you cut out the first four frequencies, you can set up another cascade with filters at 300, 360, 420, and 480 Hz. Just keep going through the harmonics until you've eliminated the noise.

Finally, after eliminating the various types of noise from your recording, you may need to brighten up the audio, because removing noise can dampen the higher frequencies and make the recording sound dull. When you boost higher frequencies, however, you usually boost noise along with it. If you own Sonic Foundry's *Noise Reduction* plug-in, you can use a cool trick to brighten higher frequencies without boosting the noise level. After you capture a noise print, set the Reduce Noise By parameter to zero. Next, activate the High-Shelf parameter, and set the frequency to about 7 kHz and the gain to about 6 dB. Finally, hit the Preview button. You'll notice that your audio now has a lot more presence, but the noise level is still low. Arboretum's *Restoration-NR* has a similar high-boost and high-cutoff control that preserves or reinforces the upper spectrum.

BLESSED SILENCE

As I mentioned earlier, you may not get rid of all the noise in your recordings, but with today's technology you can get very close to that goal without doing serious damage to your source material. It would be great if there were some magical procedure that worked for every piece of music with any kind of noise in any situation. Unfortunately, there isn't at this time.

Nonetheless, if you take your time, follow the procedures outlined in this article, and do a little experimenting, you'll find that cleaning up your recordings isn't overly difficult once you get the hang of it. Just remember that the quality of the initial source material is paramount to the outcome. If you have to leave a bit of residual noise in the signal, it's not always a tragedy. Be sure to use these powerful noise-reduction tools judiciously, and avoid overprocessing your recordings. It can be a tricky balancing act, but the results are well worth the effort.

Author, musician, and multimedia guru **Scott R. Garrigus** *(www.garrigus.com) is a frequent contributor to* **EM**. *He also publishes his own e-zine called Comp-media (www.garrigus.com/comp-media).*

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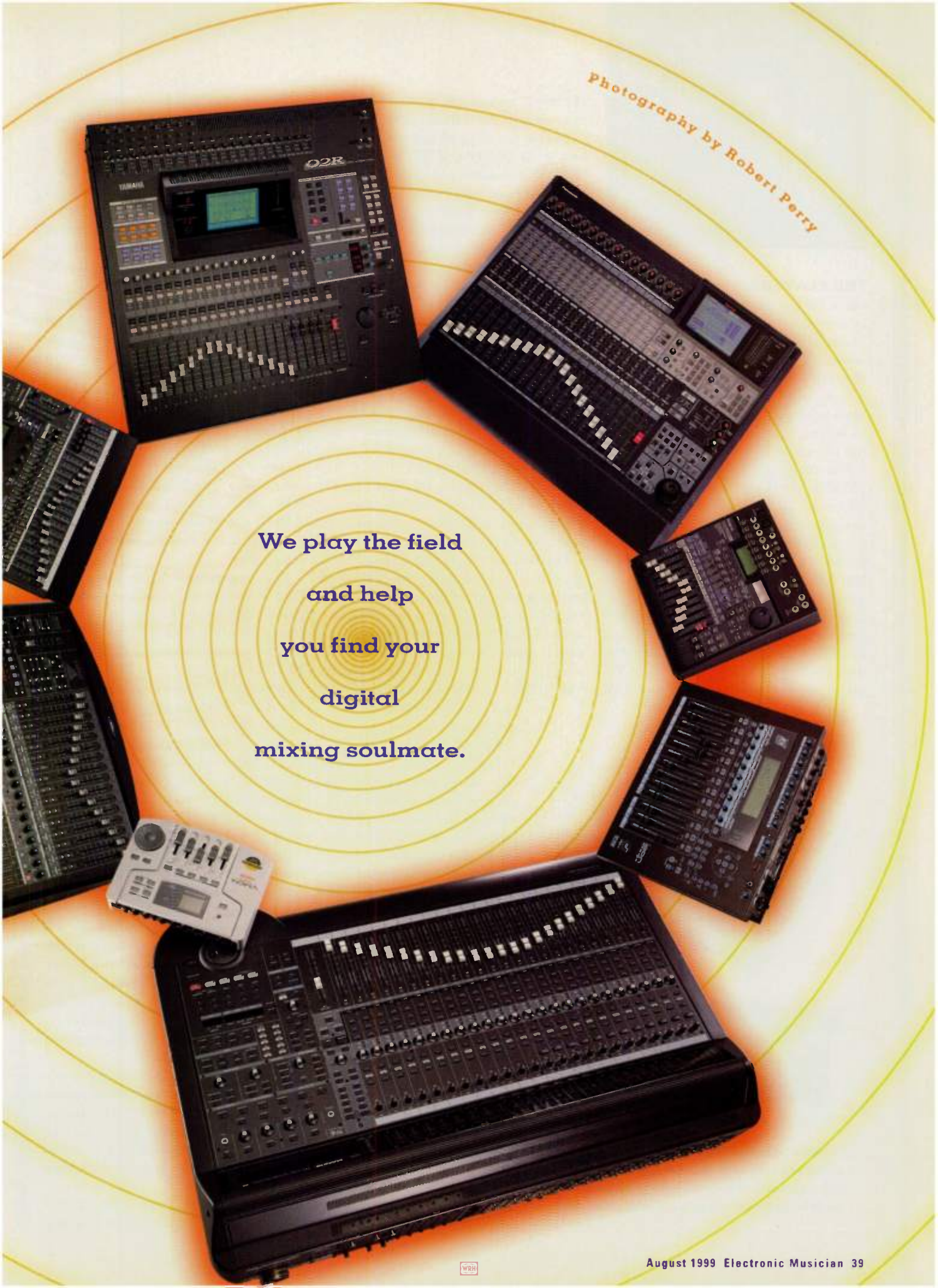
DIGITAL OR ANALOG?

By Jeff Casey

A good friend of mine who works as a post-production engineer in New York City recently told me that he declined his company's offer to install a Neve 9098 console in his studio in favor of keeping his digital mixer. I was floored; I love the gear in my studio, but I'd have a hard time turning down a classic Neve board. His explanation was simple: "I don't want the hassle of working with an analog board."

I couldn't argue with my friend's logic. Digital mixers are very flexible and powerful tools, they include a variety of signal-processing features in a compact unit, and they can interface easily with other digital gear. To boot, thanks to advances in DSP and converter technology, the best digital mixers sound almost as good as some high-end analog boards. To top it off, these units are reasonably priced—cheaper than automated analog boards as well as some unautomated, personal-studio-class analog boards with comparable signal-processing capabilities (if you include the cost of outboard gear).

With demand increasing, manufacturers are striving to build more innovative, cost-effective, and user-friendly units of all sizes, and the market has recently been inundated with new offerings. Let's compare the current crop of digital mixers and discuss how each affects life in the personal studio.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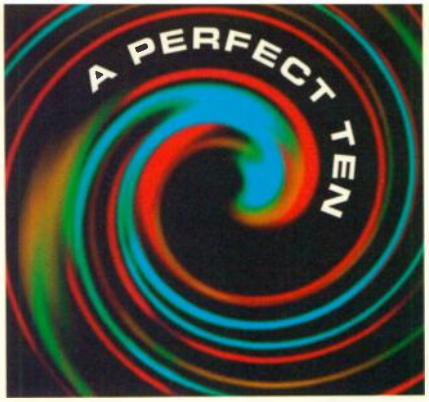

THE PLAYERS

Since **EM** last examined digital mixers as a group (see "Studio Heart Transplants" in the February 1998 issue), the market has expanded nearly threefold. Many of the boards discussed then are still around today; others that had not been released at that time have made it to stores; and one, the Korg 168RC, has been discontinued.

Not surprisingly, the Yamaha 02R ($8,899) is still one of the leaders. For all intents and purposes, the 02R is the console that started this revolution. (True, Yamaha released several digital consoles before the 02R, notably the DMP7D and the ProMix01, but these mixers were less than compelling because they lacked sufficient digital I/O, among other things.) The 02R has found a home in post-production studios around the world. Also from Yamaha are the 03D ($3,699) and 01V ($1,999), both of which were designed to target personal-studio owners.

Tascam has stopped aggressively marketing its higher-end TM-D4000 and TM-D8000 consoles in order to focus on the smaller, lower-priced TM-D1000 ($1,299), which was designed primarily for the MDM-based personal studio. Spirit by Soundcraft has been making recording consoles for years, so it's no surprise that it's in this game, too, with the Digital 328 ($4,995). The well-received Panasonic/Ramsa WR-DA7 ($4,995) has been shipping for about a year now, and the long-awaited Mackie Digital 8-Bus, or D8B ($9,999), arrived in stores last year and recently received a major operating-system upgrade.

Two of the newest digital mixer offerings are from Fostex and Roland. The Fostex VM04 ($299) holds the record for being the smallest digital mixer in production, weighing slightly less than a pound—in fact, the manual is larger than the mixer.

The Roland VM-3100Pro ($1,295) is also fairly small. Roland shipped me a unit that was one of the very first to arrive in the United States, so I feel honored to have been able to check it out. (A less expensive version, the VM-3100, will also soon be available. It lacks the VM-3100Pro's multichannel digital I/O and provides only one effects processor.)

The Allen & Heath DP1000 ($1,595) is a different animal from all the others. Although each of these mixers can be used in live-sound applications, the DP1000 is the first affordable digital console designed specifically for this task. The model I tested has two built-in 100-watt power amps. (An unpowered version, the DL1000, is also available for $1,395.)

Some of the products I wanted to evaluate were unavailable to me. The Roland VM-7100 and VM-C7200, Fostex VM200, and Generalmusic Falcon had not yet officially shipped as of this writing; at most, one or two demo models existed in the United States. As noted earlier, Tascam has been downplaying its TM-D4000 and TM-D8000, and no units were available for review. However, you will find information about these consoles in sidebars throughout this article.

The Allen & Heath DP1000 is the first digital mixer designed expressly to handle live-sound applications. Offering a wealth of features for the gigging musician, the DP1000 even has two onboard 100W power amps.

GOING THROUGH CHANNELS

Let's begin by examining each board's input channels and associated connections. After all, there's no sense falling in love with a mixer only to find out later that it doesn't provide enough inputs for your needs.

All connections on the DP1000, VM04, 02R, 03D, WR-DA7, and D8B are on the rear panels. (This is an interesting choice for the Allen & Heath board because sound-reinforcement boards often have top-panel jack fields.) The Digital 328, 01V, VM-3100Pro, and TM-D1000 provide top-panel analog input connections, which could be handy if you don't have a patch bay in your studio. The WR-DA7, Digital 328, D8B, and VM-3100Pro deliver 24-bit A/D/A conversion; the other consoles convert to and from analog at 20-bit resolution.

The largest unit of the group, the D8B is a 48-channel mixer designed like a traditional split console, with 24 input channels and 24 tape returns. Input channels 1 through 12 offer both XLR mic-level and ¼-inch TRS line-level inputs. Each of these channels has a Mic/Line switch for selecting the input source. Channels 13 through 24 are fed from ¼-inch balanced line-level jacks. The tape returns (channels 25 through 48) are accessed through I/O expansion cards installed in any of four available slots. (I'll discuss expansion capabilities later.)

The 02R is a 40-channel mixer. Both XLR and ¼-inch TRS inputs are provided for channels 1 through 8, and there are handy A/B switches on the top of the unit for selecting the source on each of these channels. Channels 9 through 24 have ¼-inch TRS inputs, with inputs 17 through 24 paired as stereo channels. The remaining channels (25 through 40) are accessed though I/O cards, which can be installed in any of the four expansion slots.

The WR-DA7 is a 32-channel console that comes standard with 16 analog inputs: 8 XLR and 8 ¼-inch TRS. All analog inputs can operate at mic or line level. The other 16 inputs are accessed by installing I/O cards in the three available slots.

The Digital 328 is also a 32-channel console, with 16 XLR mic inputs and

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16 ¼-inch TRS line inputs on channels 1 through 16. The Digital 328 comes standard with two TDIF and two ADAT Optical (Lightpipe) digital I/O ports that feed channels 17 through 32, which are the tape returns. There are no I/O expansion card slots, but considering the abundant I/O that comes on the stock unit, they're really not necessary.

The 03D has 24 mixdown channels. Channels 1 through 8 offer both XLR and ¼-inch TRS inputs, channels 8 through 16 offer ¼-inch TRS inputs only, and channels 17 through 24 are addressed through the single I/O expansion card slot. The 01V is a 24-channel console, as well. Channels 1 through 12 provide XLR and ¼-inch TRS connections, and channels 13 through 16 have ¼-inch TRS inputs that feed signals to the two stereo channels. Channels 17 through 24 are accessed by installing an I/O card in the single slot.

The VM-3100Pro offers 20 mixdown channels. (The lower-end VM-3100 has 12 channels.) Channels 1 and 2 each provide XLR mic-level and ¼-inch TRS line-level inputs. Channels 3 through 8 offer ¼-inch TRS inputs, which can be either mic or line level. Channels 9/10 and 11/12 are stereo pairs, and they are fed from RCA jacks. Channels 13 through 20 are reserved for the optional 8-channel digital I/O breakout box.

The TM-D1000 is a 16-channel mixer. Both XLR and ¼-inch TRS inputs are provided on channels 1 through 4, and ¼-inch TRS inputs are offered on channels 5 through 8. (The latter channels can be set for either mic- or line-level operation.) The unit comes standard with one TDIF port for addressing channels 9 through 16. A single expansion slot is also provided.

The DP1000 is an 8-channel mixer that provides both XLR and ¼-inch TRS inputs on channels 1 through 6. Channels 7 and 8 are stereo channels and offer left and right ¼-inch TRS and RCA inputs (which can be summed), as well as mono XLR inputs.

Finally, the VM04 is a 4-channel mixer. It has ¼-inch mic/line inputs for channels 1 and 2 and ¼-inch line-level inputs for channels 3 and 4.

INSERTS, TRIMS, & PHANTOM

The D8B, 02R, Digital 328, TM-D1000, and WR-DA7 all include ¼-inch TRS insert jacks for every XLR input. (The Digital 328 also has inserts for the ¼-inch mic/line jacks.) The 03D offers inserts on only the first two channels, and the 01V, DP1000, VM-3100Pro, and VM04 don't have external insert points.

All models except the VM04 provide top-panel trim controls for attenuating the analog inputs. The VM04 has rear-panel controls for adjusting the two mic-level inputs. In some cases, as with the WR-DA7, VM-3100Pro, and 02R, the trim controls are used to adjust between mic- and line-level operation. All three Yamaha consoles have switches for applying a pad to the mic inputs, and the Digital 328 provides a low-frequency filter at 100 Hz on every analog input channel.

Phantom power is available on all of the mixers I tested except the VM04. The 02R, 03D, and D8B all provide individual phantom power switches for each channel, but annoyingly, the 03D's and the D8B's are on the rear panel. The WR-DA7 also allows you to engage phantom power individually per channel through the software—an excellent feature. The TM-D1000, Digital 328, and DP1000 all have global phantom power buttons. The 01V offers phantom power in groups of eight channels, and the VM-3100Pro provides continuous phantom power on both XLR inputs.

GETTING OUT

The 02R, 03D, D8B, and Digital 328 deliver L/R analog output on XLR and RCA jacks and digital output on coaxial (S/PDIF) and XLR (AES/EBU) jacks. In addition, the Digital 328 provides an auxiliary stereo optical output, which can be in ADAT or S/PDIF format. The 01V has stereo analog outputs on both XLR and RCA jacks and digital outputs on coaxial RCA jacks. The WR-DA7 has only XLR and ¼-inch TRS analog outputs—there are no digital outputs on the stock console.

The TM-D1000 has ¼-inch TRS and RCA analog outs and AES/EBU and coaxial digital outs. The VM04 provides ¼-inch TRS outputs and an optical output. The VM-3100Pro has two ¼-inch TRS analog outputs, as well as coaxial and optical digital outs. In addition to the speaker-level outputs on the DP1000, there are ¼-inch L/R main outputs as well as RCA record outs, a mono output, and a Foldback Out jack.

EXPANDING YOUR WORLD

As mentioned earlier, the D8B, 02R, 03D, 01V, WR-DA7, and TM-D1000 each have rear-panel expansion card slots so you can add I/O ports. The VM-3100Pro has an optional breakout box that can be connected to the multipin port on the rear of the unit; it includes both ADAT Lightpipe and TDIF digital I/O. Each of the expansion cards provide eight channels of additional I/O, and you can install as many cards as there are available slots.

The 02R and 03D accept Yamaha-format YGDAI cards. Separate TDIF and ADAT Optical cards are available for the 01V, 02R, 03D, and WR-DA7. The D8B has a single card that provides

The smallest mixer of the bunch, the Fostex VM04 provides four channels of automated mixdown. You get a multi-effects processor, and there is EQ on each channel, but the VM04 does not support MIDI.

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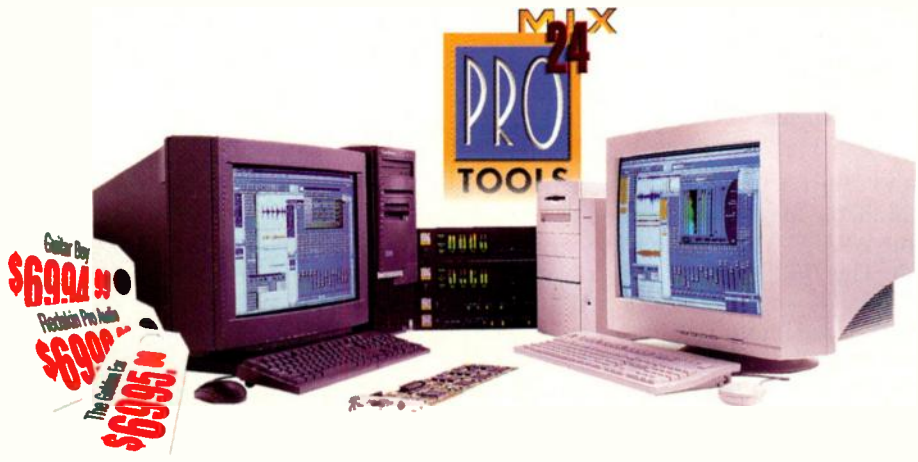

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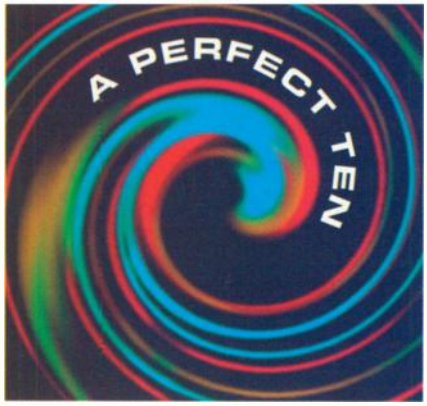

both TDIF and ADAT digital I/O. AES/EBU cards are available for each of the consoles, and the WR-DA7's card also has S/PDIF I/O. The D8B card can do real-time sample-rate conversion on every AES/EBU stereo input.

Eight-channel A/D and D/A expansion cards provide additional analog I/O for the WR-DA7, D8B, 02R, and 03D, and the 01V has 4-channel D/A and A/D cards. Tascam offers only one expansion card for the TM-D1000, which provides eight channels of additional TDIF, stereo AES/EBU, and stereo S/PDIF I/O.

ON THE SURFACE

Although each mixer is designed differently, all are easy to work with, and I didn't have much trouble finding my way around any of them. The D8B's design most resembles that of an analog console, with several dedicated knobs and buttons.

The WR-DA7, 02R, and 03D have large backlit LCD screens. These 5 × 5–inch displays show a wealth of information. The 01V and DP1000 both have slightly smaller, horizontal LCD screens. The TM-D1000, Digital 328, VM-3100Pro, and VM04 have rather small LCD screens; the Digital 328, though, provides a good number of surface controllers, so you won't need to use the display very often. All of the mixers with larger screens have contrast knobs, and each provides a Channel View page that displays most of the selected channel's information on one screen.

The D8B's horizontal LED display is somewhat small considering the console's power. However, you can connect the mixer to an IBM-compatible monitor, keyboard, and mouse, which makes it much easier to work with. (I'm a Mac user, so I wasn't able to test that setup.)

For navigating the displays, all of the consoles except the D8B offer cursor controls (Up, Down, Left, and Right). The D8B has two arrows in the EQ/effects section for scrolling through presets and parameters. The WR-DA7, 02R, 01V, 03D, and VM-3100Pro all have data-entry wheels, while the TM-D1000 and Digital 328 have smaller data-entry knobs. The 03D allows you to connect a mouse, eliminating the need to use the cursor buttons. Interestingly, the 01V and 02R do not have this feature.

To conserve physical space on the digital mixers, individual channel controls have been replaced with global controls that can be applied to each channel as needed. The global sections vary in design; however, with the exception of the TM-D1000, VM04, and 03D, each has dedicated knobs for controlling the EQ.

The Yamaha consoles rely largely on the LCD screens and cursor controls for adjusting parameters. Buttons next to each display allow you to access various pages; for example, dynamics, pan/routing, and aux-send pages can be brought up with a single button push. Navigating the screens with the cursor controls and parameter wheel can sometimes be a bit cumbersome, although using a mouse with the 03D makes a world of difference. The 03D also has four User buttons in the global section that you can assign to any number of functions.

In contrast, the WR-DA7's global-control section has dedicated knobs and buttons for assigning EQ, pan/bus, dynamics/delay, and aux sends. The 02R provides a virtual control section that handles pan, EQ, and routing assignments. The DP1000 offers a horizontal channel strip with controls for setting EQ and aux-send levels (these double as "soft" knobs—see below), and the Digital 328 has a similar strip with knobs for controlling EQ, aux send, effects send, and pan. These strips are strikingly similar to a channel strip on an analog console, and they really make you feel as though you're working on an analog board. I just wish the strips were vertical, not horizontal. (What can I say? I'm an old-fashioned guy.) Finally, the D8B offers a Fat channel strip where all DSP-based processing (including EQ) is controlled.

Each mixer has Channel Select buttons

The 01V is the simplest and least expensive of Yamaha's three digital consoles. It offers one expansion card slot for adding multichannel digital I/O.

Sporting a strikingly British look, the Spirit by Soundcraft Digital 328 offers a horizontal channel strip that provides control over EQ, pan, and aux assignment. In addition to 16 XLR and ¼-inch TRS analog inputs, the Digital 328 comes standard with two sets of TDIF and two sets of ADAT Lightpipe I/O ports.

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for assigning channels to the controls. These buttons either illuminate or flash when a channel is selected. The D8B and the 02R also have large LED displays that remind you which channel is selected, and all of the consoles with large LCD screens display the channel number somewhere on the screen.

The D8B and DP1000 have "soft" knobs on each channel, meaning that their functions change depending on the global mode. For example, the D8B's soft knobs can control pan position, aux-send level, or solo level. Similarly, the VM-3100Pro and TM-D1000 each have four soft knobs just below the displays in their global sections. Values of various parameters appear in the displays above their associated knobs. The four EQ knobs on the D8B also serve double duty in setting various effects and dynamics-processing parameters.

All the mixers except the VM04 and the VM-3100Pro have separate master left and right LED meters; the Fostex and Roland boards display master levels within the LCD screens. The D8B and Digital 328 each provide LED meters for the input channels, and the WR-DA7 and 02R have optional meter bridges. The WR-DA7, 02R, 03D, 01V, VM-3100Pro, and DP1000 have software pages that display input-level meters. I prefer to have hardware LED meters, though their absence wouldn't stop me from buying any one of these mixers. Most of the higher-end consoles permit you to view other levels, such as bus and aux-send levels, using the channel meters.

The brand-new VM-3100Pro from Roland packs 20 mixdown channels, dynamics and multi-effects processing, and a multichannel digital I/O option into a sleek-looking unit.

In addition to the channel-level meters, the 02R, TM-D1000, and WR-DA7 offer peak LED indicators, and the 02R and WR-DA7 also have signal-present LEDs. The D8B's soft channel knobs, and the knobs on the Digital 328's horizontal channel strip, each have circles of LEDs around them that illuminate so that you can quickly note the level of each, which I think is pretty cool.

Except for the VM04, each console has space for console marking tape. Of course, few people will have trouble remembering what's routed to the VM04's four channels. Some consoles provide space beneath each channel for writing directly on the console without using marking tape. I like the fact that the WR-DA7 and D8B offer separate strips for marking the inputs as well as the faders. The 02R has an additional strip for labeling the tape returns.

FADING FAST

The TM-D1000, VM04, and DP1000 are the only consoles I tested that have dedicated faders for each channel; the rest use fader layers to conserve space. Fader layers allow a single fader to control multiple volume levels. For example, the Digital 328 is technically a 32-channel mixer, but there are only 16 faders on its surface. Here's how it works: fader layer 1 controls inputs 1 through 16, fader layer 2 controls tape

ROLAND VM-7100 AND VM-7200

The flagship of Roland's V-Mixer line will be the VM-C7200 V-Mixing Console ($3,695), which can handle 94 channels of automated digital mixing. The VM-C7200 features 24 motorized channel faders in addition to a stereo master fader. The mixer's FlexBus allows up to 12 bus settings to be saved, and an EZ Routing feature lets you save and recall common settings. Scene and dynamic memory are also provided. Roland's other 94-channel V-Mixer, the VM-C7100 ($2,995), is similar to the VM-C7200 but features just 12 channel faders and a master fader.

The VM-C7200 and its sibling are actually control surfaces for the rack-mountable VM-7200 48-channel V-Mixing processor ($2,795) and for a 38-channel version, the VM-7100 ($1,995). The processors are the workhorses of the system. The VM-7200 houses 40 analog input channels (20 ¼-inch TRS and 20 XLR), 8 bus outputs, 8 assignable analog outputs, and AES/EBU digital I/O. The VM-7100 has 10 XLR and 10 ¼-inch TRS analog inputs. The systems also come with 24-bit A/D and D/A converters.

Each channel has dynamics processing and 4-band parametric digital EQ. You get two assignable stereo effects processors and one stereo master effects processor. Each VM processor comes with a built-in VS8F-2 stereo effects expansion board that generates various multi-effects; three additional VS8F-2 boards ($395 each) can be user installed per processor, for up to 8 stereo or 16 mono effects.

The VM processors are controlled by a VM-Link interface, which sends data and audio from the mixing consoles through standard AES/EBU cables. Using the VM-Link, the processor can be controlled by the mixing console at distances of up to 220 yards. You can use either VM processor with either console, and the processors can be cascaded together. In addition, one VM processor can be controlled by two cascaded mixing consoles, expanding the number of physical faders. Both the VM-7100 and VM-7200 are completely MIDI-controllable and can be automated from a Mac- or PC-based software sequencer.

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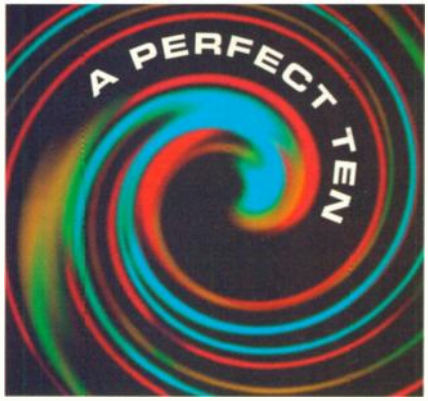

returns 17 through 32, and fader layer 3 controls the eight buses and six aux sends. The configuration of the fader layers varies from console to console.

The bus masters are usually incorporated into a channel-fader layer, but the TM-D1000 and WR-DA7 have dedicated bus faders. The WR-DA7 offers only four bus faders, despite the fact that there are eight buses, so the dedicated bus faders each have two fader layers. The WR-DA7, VM-3100Pro, and D8B allow you to send MIDI control messages with the faders, as well.

Fader layers are selected by pressing dedicated buttons on the console. Doing this generally changes the fader layers globally, though the WR-DA7, D8B, and Digital 328 let you swap individual faders between the inputs and tape returns. When you're working on a mix, it's important to know which fader layer is active, and all of the consoles do a good job of keeping you informed, using either illuminated buttons or LEDs.

The Digital 328, WR-DA7, 01V, 02R, 03D, and D8B have motorized faders. Although none of the motorized faders feel "cheap," by any means, those on the Digital 328 and WR-DA7 are probably the smoothest of the bunch. The larger consoles (WR-DA7, Digital 328, D8B, and 02R) have 100 mm long-throw faders, as does the DP1000. The 03D, 01V, TM-D1000, and VM-3100Pro offer medium-throw faders, and the VM04 has miniature faders. The consoles that don't feature motorized faders have fader-nulling features for synching the hardware faders with the automation software.

ROUTE ME!

Panning channels is done differently on each console. The TM-D1000 is the only unit that offers dedicated pan controls for each channel, and they are all detented at the center position—a very nice touch.

With the WR-DA7, Digital 328, 01V, and 02R, panning is accomplished by selecting a channel and then adjusting an infinitely rotating, global pan knob. The 02R has a small LED display next to the pan knob that illustrates the pan position, and as mentioned earlier, the Digital 328 has a circle of LEDs around the knob. The D8B and DP1000 use the soft channel knobs for altering pan position. The global soft knobs take care of panning on the VM-3100Pro, which is a bit hard to do because the

DIGITAL MIXER SPECIFICATIONS

Product	Mixer Configuration	Mic Inputs (XLR)/Line Inputs	# and Type of Digital Inputs	# and Type of Digital Outputs	# and Type of Option Card Slots	Analog Inserts/ Direct Outs	# of Analog Aux Sends/Returns	A/D/A Converters	EQ Type	# of Dynamics Processors
Allen & Heath DP1000	8 x 2	8/8	0	0	0	0/0	1/1	20-bit	4-band parametric	8
Fostex VM04	4 x 2	2 (¼")/4	0	(1) S/PDIF (optical)	0	0/0	0/0	20-bit	2-band shelving	0
Mackie Digital 8-Bus	48 x 8 x 2	12/24	(1) S/PDIF; (1) AES/EBU	(1) S/PDIF; (1) AES/EBU	(4) I/O; (3) DSP; (1) sync	12/0	12/16	24-bit	4-band parametric	48
Panasonic/Ramsa WR-DA7	32 x 8 x 2	8/8	(1) AES/EBU or S/PDIF	(1) AES/EBU or S/PDIF	(3) I/O; (1) SMPTE/ V-Sync	8/0	4/4	24-bit	4-band parametric/2-band parametric on aux returns	42
Roland VM-3100Pro	20 x 8 x 2	2/12	(1) S/PDIF	(2) S/PDIF	(1) I/O	0/0	2/2	24-bit	3-band	2
Spirit by Soundcraft Digital 328	32 x 8 x 2	16/16	(1) AES/EBU; (1) S/PDIF; (2) TDIF; (2) ADAT Optical	(1) AES/EBU; (1) S/PDIF; (2) TDIF; (2) ADAT Optical	0	16/16	4/5	24-bit	3-band parametric	2 floating
Tascam TM-D1000	16 x 4 x 2	4/8	(1) TDIF; (1) AES/EBU; (1) S/PDIF	(1) TDIF; (2) AES/EBU; (2) S/PDIF	(1) I/O; (1) effects	4/4	4/2	20-bit	3-band, parametric mid	4 floating
Yamaha 01V	24 x 4 x 2	12/12	(1) S/PDIF	(1) S/PDIF	(1) I/O	0/0 (4 using Omni outs)	4 (using Omni outs)/2	20-bit	4-band parametric	22
Yamaha 02R	40 x 8 x 2	8/24	(1) AES/EBU; (2) S/PDIF	(1) AES/EBU; (2) S/PDIF	(4) I/O; (1) effects	8/0	6/2	20-bit	4-band parametric	40
Yamaha 03D	24 x 4 x 2	8/16	(1) AES/EBU; (1) S/PDIF	(1) AES/EBU; (1) S/PDIF	(1) I/O	2/0	4/2	20-bit	4-band parametric	36

knobs are small and sensitive and often don't stop where you want them to. The 03D and VM04 use the data-entry knob to change pan value.

The Digital 328, WR-DA7, D8B, and 02R are 8-bus mixers (that is, they have eight subgroups, not including aux sends and monitor outs). The 03D, 01V, and TM-D1000 each have four buses, and the VM04 and DP1000 have two. Assigning buses from the D8B, WR-DA7, 02R, TM-D1000, and Digital 328 is straightforward: you select a channel, go to the dedicated bus-routing section, and select one or more destination buses. With the 03D and 01V, you need to access the pan/routing page onscreen and make assignments from there.

Roland's approach to buses is different from that of the other manufacturers. The VM-3100Pro has 12 identical buses, the levels of which are controlled from eight soft knobs. These buses can be assigned pre- or postfader and pre/post-EQ, and they can also be used as aux sends, monitor sends, or subgroups. Roland calls the VM-3100Pro a 20-input × 8-group × 2-output console, with four external aux sends. The architecture's routing versatility is undeniable, and if you make eight of the buses postfader/EQ, you can indeed use them as subgroups. However, controlling subgroups from soft knobs is clumsy; I prefer using faders for this purpose. Of course, soft knobs are as good as traditional knobs for controlling aux and monitor sends because you don't constantly tweak these levels during a mix.

I had little trouble finding my way around any of the consoles.

Assuming your mixer is properly connected to your multitrack recorder, some of the consoles, including the D8B, the WR-DA7, and the three Yamaha boards, allow you to route channels to direct outputs, bypassing the busing section altogether. Of course, all of the consoles let you route channels to the master L/R bus.

The stock WR-DA7 and 02R provide no analog or digital bus outputs; you have to install I/O cards in order to use the buses to send signals externally. Likewise, the Digital 328 provides no analog bus outputs, though you can use the provided TDIF or Lightpipe ports for this purpose. Unless you're working with an analog multitrack, this shouldn't be a problem.

The 01V provides four Omni Outs that can be used to output bus, aux-send,

# of Effects Processors	Automation: Scene/MIDI/Dynamic	Frequency Response	Signal-to-Noise Ratio	Total Harmonic Distortion	Channel Crosstalk (@ 1 kHz)	Phantom Power	Options	Dimensions (inches)	Price
2	yes/yes/no	20 Hz–20 kHz	N/A	<0.01%	-90 dB	yes	none	17.4 x 6.7 x 15	$1,595
1	yes/no/no	N/A	N/A	0.01% (typical)	>70 dB	no	none	10 x 2 x 7.3	$299
2	yes/yes/yes	20 Hz–20 kHz	115 dB	0.005%	-90 dBu	yes	clock card; I/O cards; effects card	27.1 x 37.6 x 8.7	$9,999
0	yes/yes/yes	20 Hz–20 kHz	-96 dB (typical)	<0.1%	-90 dB	yes	8-channel I/O cards for ADAT, TDIF, AES/EBU, and S/PDIF; analog meter bridge; SMPTE V-Sync; automation software for Mac or PC	27.5 x 13.6 x 23.7	$4,995
2 stereo or 4 mono	yes/yes/no	N/A	N/A	N/A	N/A	yes	DIF-AT ADAT/ TDIF I/O	3.75 x 11.8 x 13.5	$1,295
2	yes/yes/no	20 Hz–20 kHz	N/A	<0.005%	>95 dB	yes	TDIF-to-mic preamp; TDIF-to-analog interface; TDIF-to-AES/EBU interface	N/A	$4,995
2	yes/yes/no	20 Hz–20 kHz	80 dB	<0.1%	>90 dB	yes	IF-TAD; IF-TD1000; FX-D1000; MA-AD8; IF-AE8; RM-D1000	17.3 x 4 x 15	$1,299
2	yes/yes/no	20 Hz–20 kHz	98 dB	<0.1%	-70 dB (ch. 1–12); -60 dB (ch. 13–16)	yes	digital I/O cards: TDIF, ADAT, AES, analog input	16.9 x 5.8 x 20.4	$1,999
2	yes/yes/yes	20 Hz–20 kHz	105 dB	<0.2%	-70 dB	yes	meter bridge; digital I/O: up to 32 TDIF, AES/EBU, ADAT, or Yamaha w/option cards	26 x 8.5 x 2.7	$8,899
2	yes/yes/yes	20 Hz–20 kHz	105 dB	>0.01%	-70 dB	yes	I/O interface cards: TDIF, ADAT, AES/EBU, Yamaha; cascade kit; rack ears	18.1 x 20.3 x 8	$3,699

channel direct output, or master L/R signals to the analog world. Similarly, the 03D has four dedicated analog bus outputs (¼-inch TRS). With either console, buses can be routed to the digital I/O card. The TM-D1000 provides four ¼-inch TRS and RCA analog bus outputs. In addition, the buses can feed the TDIF port.

The WR-DA7 offers six aux sends: two S/PDIF and four analog. The Digital 328 provides four external sends and two internal, which are routed to the onboard multi-effects processors (discussed later). The 02R has eight aux sends, two of which are internally routed to the onboard effects. The 03D has four external and two internal sends, as does the 01V.

The TM-D1000's four buses double as aux sends when the console is in Mixdown mode, and they can be routed to the external outputs or to the internal effects. (See the review in the July 1999 issue of **EM**.) The D8B offers eight bus outputs and a whopping 12 aux sends, which can be used for external effects, internal effects, cue mixes, and more. The DP1000 has one external aux send and two internal sends. The VM04 provides one internal send for the effects and has no external sends.

Here again, some of the consoles provide global knobs for setting aux-send levels, while others rely on channel soft knobs. I like Yamaha's arrangement, which uses the channel fader for setting aux-send levels. Even better, the WR-DA7 offers both fader and global control, and the 02R has fader and

GENERALMUSIC FALCON

Generalmusic recently unveiled the Falcon, the first in the company's LEM line of digital mixing consoles. Although we had hoped to get a unit for testing, the first units had not yet reached the United States when we went to press.

The Falcon offers 12 channel inputs, six of which are XLR mic-level, as well as AES/EBU and S/PDIF I/O. (An optional Lightpipe interface is also available.) You get individual channel EQs, reverb, and standard multi-effects, along with 12 dynamic insert effects (including delays), which can be set up in several different ways, such as in series or parallel, and routed to any of the input channels. There are user libraries for all of the effects and EQs. A 4-band graphic EQ is offered on each of the two stereo inputs. For external processing, four aux sends are available.

The Falcon includes scene and dynamic automation and has 60 mm motorized faders. Two or more consoles may be cascaded. Especially noteworthy is the fact that this is the only digital mixing console that has dedicated controls (pan, aux-send levels, and so on) for each channel.

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soft-knob control over aux-send levels.

The external sends, and in some cases the internal sends, can be designated pre- or postfader. All of the mixers have dedicated aux and effects returns, though the D8B's configuration is clearly the most complex, providing 16 internal effects returns and eight aux returns.

TAKE A LISTEN

All of the mixers offer solo functions. (The DP1000's solo feature sends signals to the headphone output rather than to the traditional monitor mix, but again, this board is meant for sound-reinforcement applications.) The WR-DA7, Digital 328, 01V, DP1000, and D8B have dedicated solo buttons; on the rest of the consoles, other buttons—Channel On on the 02R and 03D, Channel Select on the VM04 and VM-3100Pro, and Mute on the TM-D1000—double as solo buttons when the global Solo mode is engaged.

The D8B, WR-DA7, and 01V have solo indicators that flash when a channel is soloed—a nice feature that will keep you from racking your brain trying to figure out why no sound is coming out of your monitors. Most of the consoles allow you to designate solo as AFL or PFL.

Although all of the mixers have channel mutes, the DP1000, Digital 328, D8B, and TM-D1000 are the only consoles with buttons that are actually labeled "mute." The Yamaha consoles and the WR-DA7 have Channel On buttons; with the VM04 and VM-3100Pro, the Channel Select buttons double as mutes.

The WR-DA7, 01V, 02R, 03D, and D8B offer mute grouping, though with scene automation this feature is important only if you want to mix on the fly. All five of these mixers have a fader-grouping feature, as does the VM-3100Pro.

The WR-DA7, D8B, 02R, TM-D1000, and Digital 328 provide comprehensive monitoring sections, with controls for selecting monitoring source (Master L/R, 2-Track A, Aux 1/2, and so on) and setting the levels of the various control-room and studio-monitor sends. The Digital 328's monitor section is located near the top of the console, and it's mixed in with some other knobs and buttons, so it's not as accessible as the others.

Except for the low-end VM04 and live-oriented DP1000, each mixer has a knob for adjusting the control-room level. The WR-DA7, Digital 328, 02R, TM-D1000, and D8B have Mono buttons, which I consider a very important feature. The Digital 328, 02R, and D8B also offer handy Dim buttons for lowering the monitor level by a preset amount. The WR-DA7, 02R, and D8B have built-in talkback microphones; in the D8B's case, the mic can be routed to any of the aux sends or buses.

The WR-DA7 provides Monitor A and Monitor B sends, and the D8B and 02R have dedicated monitor sends. The D8B has buttons labeled Main and Near-Field for selecting control-room speakers and is the only console I tested that offers dedicated cue sends—two of them—which can be taken from auxes 9/10 or 11/12. All of the consoles have headphone jacks, and all except the VM-3100Pro have headphone volume knobs.

I SHALL RETURN!

Every mixer except the VM04 has at least one dedicated 2-track input. The 03D provides analog 2-track input on RCA and ¼-inch jacks, as well as digital input on coaxial jacks. The Digital 328 has two analog inputs, each on ¼-inch TRS jacks, and an S/PDIF digital input. The stock WR-DA7 has one pair of ¼-inch analog inputs but no digital input. The 02R provides two analog 2-track inputs on ¼-inch and RCA jacks, as well as an AES/EBU input and two coaxial digital inputs. The 01V has one analog input on RCA, along with a coaxial input.

The TM-D1000 has RCA and ¼-inch analog inputs and AES/EBU and S/PDIF digital inputs. (With the expansion card, two more S/PDIF inputs and an AES/EBU input are added.) The D8B has three ¼-inch inputs, as well as AES/EBU and S/PDIF inputs. The VM-3100Pro does not provide a 2-track button in the monitor section, though it does have an optical S/PDIF 2-track input. With most units, the digital 2-track inputs can be routed to a number of places within the console. For instance, with the VM-3100Pro these inputs can be routed to any and all channels.

TONAL TOUCH-UPS

Internal processing on the WR-DA7, D8B, 01V, 02R, and 03D is at 32-bit resolution; the rest of the consoles operate at 24-bit resolution. All of the onboard EQs sound good, and I would have no reservations about using any of them on professional projects.

The D8B and 02R offer four bands of fully parametric EQ on every input channel and tape return. The 03D has 4-band parametric EQ on each input, subgroup output, and aux send. The WR-DA7 provides a 4-band parametric EQ for the input channels and buses, and a 2-band parametric for the aux returns. The DP1000 has four bands of parametric EQ on every channel and two 10-band graphic EQs on the master L/R bus that can be used to tune the room. The 01V provides a 4-band parametric EQ on input channels 1 to 16; channels 17 to 24 get only a 2-band EQ each.

The VM-3100Pro and Digital 328 have three bands of parametric EQ on the input channels, and the TM-D1000 provides a 3-band EQ on the input chan-

The Yamaha 02R's midlevel sibling, the 03D, provides many of the same features as its larger companion at about half the price.

For manufacturer information, please see Contact Sheet, p. 163.

nels that includes two shelving filters (high and low) and a mid-band parametric EQ. The VM04 has a 2-band EQ on each channel: high and low shelving filters set at fixed frequencies.

The 02R has the most comprehensive EQ display of the bunch: the screen shows a virtual image of the parameter knobs as well as a graphic EQ curve, and numeric displays next to the EQ controls show the value of each parameter setting (gain, frequency, and bandwidth). The D8B, if used with a computer monitor, also provides a comprehensive EQ display. The 01V, 03D, WR-DA7, VM-3100Pro, and DP1000, like the 02R, display EQ curves graphically.

As mentioned earlier, the WR-DA7, Digital 328, 01V, 02R, and DP1000 have three EQ knobs (for frequency, bandwidth, and gain), which can be assigned to any of the frequency bands. The D8B, on the other hand, offers knobs that can either control one parameter at a time for four bands of EQ, or control band-select, frequency, gain, and bandwidth for one band of EQ. The VM-3100Pro's three knobs control only the gain of the three bands; other parameters are adjusted through the display. A nice feature of the VM-3100Pro brings up the EQ screen as soon as you make a change to any of the EQ knobs.

All of the mixers let you store EQ settings in some fashion. The 01V, 02R, 03D, D8B, VM-3100Pro, and WR-DA7 let you store EQ curves in a user library as well as within a scene (snapshot); the Digital 328, DP1000, TM-D1000, and VM04 simply store EQ settings as part of the scene.

TAKING CONTROL

Dynamics processors are available on every mixer I tested except the VM04. All offer compression and limiting; in fact, that's all the VM-3100Pro provides. The rest of the mixers' dynamics processors can handle gating, as well. The WR-DA7, TM-D1000, 01V, 02R, and 03D all offer expansion, and all of the Yamaha mixers provide companding and ducking functions, too.

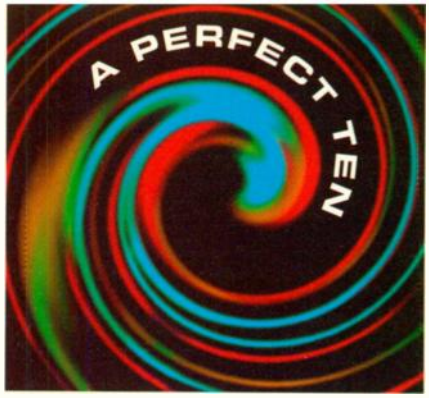
The WR-DA7, D8B, 02R, and 03D offer separate dynamics processors for every input channel, and the D8B and 02R have dynamics processing on every tape return. In addition, the WR-DA7, 02R, and 03D have dynamics processors on each bus, including the master L/R bus. The 01V and 03D offer dynamics processing on the aux sends, and the 03D offers it on the effects returns. However, the 01V has dynamics processing only on input channels 1 through 16, so you'll have to use outboard gear or do without on channels 17 through 24. The DP1000 incorporates a discrete compressor and gate on every input channel—a great feature, especially for doing live sound, which is this board's forte.

In Record mode, the TM-D1000 has four channels of dynamics processing (eight channels with the optional FX-D1000 expansion card installed), which can be applied to any input channel. The VM-3100Pro provides two compressors that you can assign to any of the input channels.

The Digital 328 has two dynamics processors that can be assigned to any of the inputs, the master L/R bus, or the effects returns. Now, you might think that this is not a lot of dynamics processing power for such a big console—and you're right. If you rely mainly on onboard processing, this could be a serious limitation. However, if you're like me and prefer to use higher-end outboard analog compressors, the limited dynamics processing on the Digital 328 isn't a major concern.

The higher-end consoles provide the full range of dynamics parameter controls, including attack and release settings, while the lower-priced mixers offer a more limited parameter set. All of the Yamaha consoles, along with the WR-DA7 and D8B (when used with a computer monitor), provide graphic displays for the dynamics processors; the WR-DA7 is the only console with a dedicated set of controls for dynamics processing. I particularly like the fact that the DP1000 displays any given parameter for all eight dynamics processors simultaneously—a feature conducive to live-sound situations where you might need to adjust two or more compressors or gates at once. The Yamaha consoles, the WR-DA7, and the D8B all offer gain-reduction meters for the compressors. The Digital 328's master L/R LED meters can be switched to display gain reduction for each of the two dynamics processors.

The VM-3100Pro, TM-D1000, Digital 328, 01V, 02R, 03D, and D8B all provide libraries for storing dynamics settings. The Yamaha consoles also come with a library of presets, which, surprisingly, aren't too bad.



EFFECTIVELY PROCESSED

Internal multi-effects processors are available on every mixer except the WR-DA7, which has just channel delay. The lack of onboard multi-effects is the only real drawback to the WR-DA7, but it's a big one.

All of the consoles come standard with two onboard processors, except the VM04, which has only a single effects processor, and the WR-DA7, which offers channel delay but no effects. TM-D1000 owners can purchase the optional FX-D1000 expansion card, which adds another two multi-effects engines (as well as another dynamics processor) to the mix. Similarly, TC Electronic makes the UnitY effects expansion card for the 02R (starting at $1,195), which gives you the equivalent of a TC Electronic M2000 outboard processor on a card.

Most of the processors provide the standard array of effects, including reverb, delay, and modulation. In addition, the VM-3100Pro generates some cool guitar effects—a nice feature, especially in combination with the ¼-inch, guitar-level input on the top panel. It also has speaker modeling and mic modeling, which is unusual in this field. Like the WR-DA7, all of the Yamaha consoles offer in-line delay on every input channel.

Mackie's effects arrangement is the most flexible of the bunch. There are four DSP expansion board slots on the rear of the D8B console, one of which has a preinstalled card. Each card houses two multi-effects engines, so you can have up to *eight* onboard processors. Third-party manufacturers are reportedly designing cards too, so the D8B will be the first digital mixing console to institute the plug-in concept. I can't wait to hear the results.

All of the mixers' multi-effects processors are user programmable, and each allows you to store edited programs in a user location. I've always been a fan of Yamaha effects, and the effects processors in the company's three digital mixers were my favorites of the whole bunch, offering a wide variety of presets and lots of editable parameters. I also liked the Digital 328's Lexicon-created effects. The D8B's stock effects were roughly third on my list: they're clean, there are plenty of editable parameters, and you get a good variety of programs, including a really nice pitch shifter/ harmony processor by IVL. Roland's effects, although solid and pleasing, were not among my favorites. The Fostex effects were the weakest of the bunch; they're usable, but not thrilling.

All of the effects are accessed by using the displays along with the mixer's cursor buttons and/or data wheel. None of the units have dedicated controls for effects parameters, except in the delay section of the WR-DA7.

Designed for the musician, the Panasonic/Ramsa WR-DA7 provides dedicated front-panel controls for EQ, dynamics, pan, and aux send and bus assignment. The stock I/O is limited, so if you buy this board you'll need to budget extra money for expansion cards.

MODERN TRANSPORTATION

The D8B, Digital 328, and VM-3100Pro have dedicated transport sections for controlling external multitracks. (The cursor keys on the Yamaha consoles and on the WR-DA7 double as transport controls.) Overall, the D8B offers the most comprehensive transport section of the bunch. The D8B and the Digital 328 have separate LEDs in the transport section that display tape position. The Digital 328, WR-DA7, and D8B come with Locate buttons, and the WR-DA7 and D8B provide looping features.

All of the mixers except the VM04 support MIDI Machine Control and can be synched to a sequencer via MIDI Time Code. The WR-DA7 has a built-in Mac-only serial interface, and the 01V, 02R, and 03D have a built-in Mac/PC interface, eliminating the need for an outboard MIDI interface. In addition, the stock WR-DA7, Digital 328, 02R, and 03D have word-clock connections, and the 03D also has a To Editor port for hooking up the mixer to a video editor. Panasonic/ Ramsa offers a SMPTE/video sync card for the WR-DA7, and Mackie offers an optional Apogee sync card for the D8B that provides word-clock I/O.

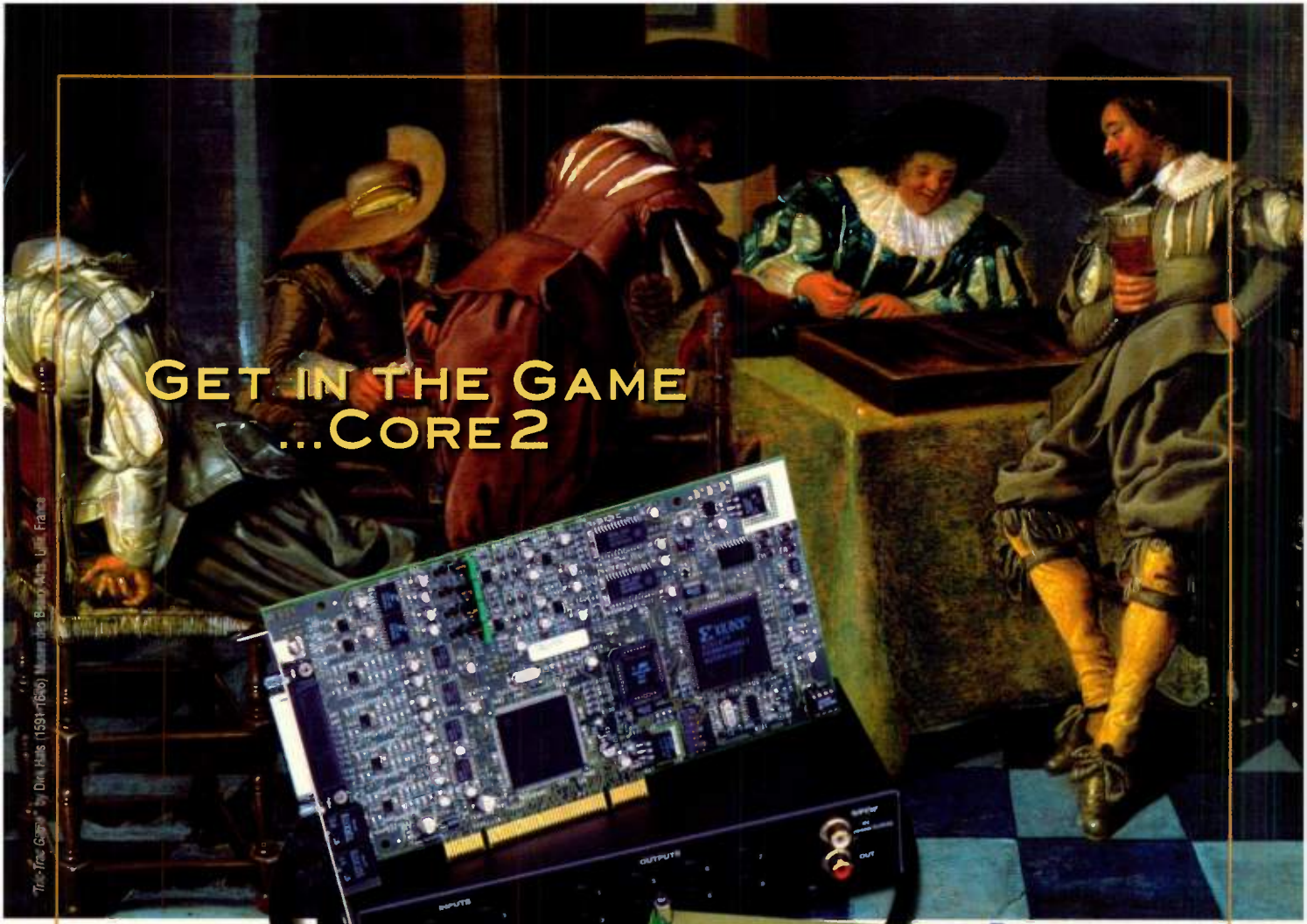
A CONTROLLED MIX-UP

With a digital mixer, control is the name of the game. Aside from such niceties as onboard dynamics processing and digital I/O, the ability to automate your mix—whether dynamically (that is, making changes in real time) or via scenes—is what really raises digital boards above their analog counterparts.

Each console offers scene automation,

FOSTEX VM200

A larger version of the Fostex VM04, the VM200 ($1,499) is an 8-channel mixer that has eight analog inputs and eight channels of ADAT Lightpipe input. Internal processing is handled at 32-bit resolution. There are eight motorized faders, which work in tandem with the snapshot automation. In addition, it offers internal multi-effects processing and a 4-band parametric EQ on every channel. The VM200 is completely MIDI compatible, and two consoles may be linked together to double your number of mixdown channels.



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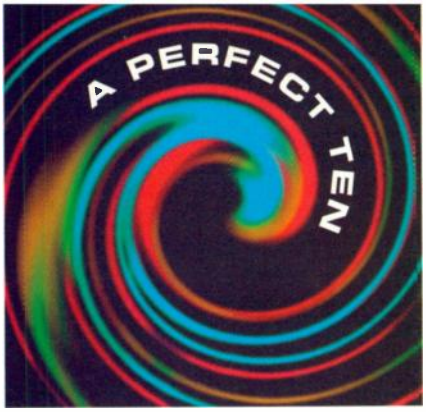

in which a snapshot is taken of all console parameters and stored in memory for later recall. All have libraries for storing scenes, and the WR-DA7, Digital 328, 01V, 02R, 03D, VM-3100Pro, and D8B offer an Undo/Redo feature. The Digital 328's Undo/Redo is especially neat: when you change any parameter on the console, the Undo button lights up in red, and when you press Undo, the Redo button lights up in green. Each mixer offers a finite number of scene memory locations, but if you are using the console with a software sequencer, you can store an infinite number of scenes on your hard drive. (The VM04, of course, cannot store scenes to a sequencer because it has no MIDI features.)

The DP1000 has a nifty Venue button that recalls the general parameters of the mixer (such as graphic EQ and master levels), allowing you to quickly customize the console for each room the second time you visit. A Song button lets you toggle easily through scenes based on your set list.

Except for the VM04, all of the consoles I tested offer real-time MIDI control over almost all parameters (exceptions include the analog trim pots on mic inputs), allowing you to automate your mix dynamically using an external sequencer. Additionally, the WR-DA7, 02R, 03D, and D8B have onboard dynamic automation, so you don't need to use a separate sequencer with them.

The automation on the D8B is the most advanced of all the mixers. This console has a separate, rack-mounted CPU that houses a large-capacity internal hard disk and a 3.5-inch floppy drive for storing lots of mix information.

ODDS AND ENDS

The WR-DA7, VM-3100Pro, DP1000, Digital 328, 01V, 02R, and 03D have internal power supplies and connect directly to a wall outlet. The VM04 uses a wall-wart adapter, and the TM-D1000 comes with an external lump-in-the-line adapter. The D8B has a large, thick cable that runs from the console to the CPU, where the power supply is housed; the CPU then connects to a wall outlet with a standard IEC cable.

Some of the smaller mixers, including the TM-D1000, VM04, VM-3100Pro, and have footswitch jacks, allowing you to use a footpedal to scroll through automation scenes, which is a fine feature if you're using these consoles at live gigs. The DP1000 has several footpedal jacks; two are used to scroll through scenes, while two others accept variable footpedals for controlling the levels of the onboard effects processors. Cool deal.

The WR-DA7, D8B, 02R, and 03D have surround-sound capabilities, supporting both 5.1 and LCRS. The D8B also supports 7.1 surround sound. I found the 03D's surround features very easy to work with, especially when using a mouse to navigate the display.

You can cascade all of the Yamaha consoles with an identical model by installing one card in either unit; you can also cascade an 02R and 03D together. You can link two WR-DA7s or two D8Bs, although you'll need to purchase two cards to accomplish this. The Digital 328 can be connected to another unit with a multipin cable—no card is necessary.

The Yamaha 02R is the oldest mixer discussed here, and it is still one of the most powerful. Featuring dedicated EQ, dynamics, and delay on every channel, the 02R also comes with powerful multi-effects, surround-sound processing, and expansion slots.

HARD COPY

Having good documentation for a machine as complex as a digital mixing console is important. The manufacturers of the ten consoles I tested succeeded to varying degrees in creating well-written and clearly organized manuals.

The documentation for the D8B is very thorough and comes in a three-ring binder. It's divided into multiple sections that are reasonably well organized, but I would have preferred one continuous manual. The included tutorial video is excellent.

The TM-D1000 comes with several small pamphlets: a reference manual, an operations manual, and a tutorial. This arrangement might be helpful to the novice user, but the lack of organization drove me crazy when I tried to find specific information. Tascam plans to include a tutorial video in the near future.

The VM-3100Pro manual is short, at 72 pages. It does a good job of explaining how to get up and running with the mixer; however, finding specific information is difficult. The DP1000 manual is also brief but does the job, explaining all of the console's unique features and how to work with the board at live gigs. Its lack of an index, though, was a pain in the neck.

All of the Yamaha manuals are thorough and well indexed. The WR-DA7 also comes with a comprehensive and logically organized 300-page manual that's probably the best of the bunch. Finally—and I hate to say it—the Digital 328's manual is awful. First of all, it's bound horizontally, so you have to flip pages up and over to thumb through it. Furthermore, it's broken down into three smaller manuals, so it's hard to know which one contains the information you're looking for. Finally, it has no index. Three strikes! It's a good thing I like this console so much.

JUST FOR YOU

Having compared the ten mixers feature by feature, let's sum up each one individually, discuss its strengths and

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weaknesses, and examine how it fits into different studio setups.

Allen & Heath DP1000. This console clearly has the most specific audience. In addition to the built-in power amp, the DP1000 has several features that make it ideal for use at live gigs. It's one of the easier mixers to use, and the four footpedal jacks allow you to operate it from the stage. The two onboard effects processors are solid, and having a dedicated compressor *and* a gate on each input channel is a major bonus.

Fostex VM04. How often do you really need a digital mixer with only four channels? Actually, although it may seem more like a novelty than a serious mixer, the VM04 has several potential applications. For instance, you could use this mixer at live acoustic gigs (say, a folk duo with vocals and a couple of guitars) or for a presentation. You could even use it as a small onstage keyboard submixer, if you're using only a couple of synths. Just remember, you'd probably pay a comparable price for an analog mixer of a similar size, so if you're in the market for a mini-mixer, you might as well go digital. For $299, how can you go wrong?

TASCAM TM-D4000 AND TM-D8000

The Tascam TM-D4000 ($4,299) is a 36-channel digital mixer with eight buses and six aux sends. A 4-band parametric EQ and a dynamics processor are available on every channel, each with a library for storing settings. The console comes with eight XLR mic inputs with insert jacks, and there are three card slots for adding TDIF, ADAT Optical, AES/EBU, and analog I/O. All conversion is 24-bit. Snapshot and dynamic automation are provided, and the TM-D4000 is fully MIDI compatible. The mixer comes with onboard multi-effects processors, as well as surround-sound mixing capabilities. The TM-D4000 also provides a transport section for controlling a multitrack. Up to four consoles can be cascaded.

Another, larger version, the TM-D8000 ($10,799), comes standard with 16 analog and 16 digital inputs (switchable TDIF, AES/EBU, S/PDIF) and 24 channels of TDIF I/O. The D8000 has eight buses and six analog aux sends, two of which can be switched to AES/EBU. A 4-band parametric EQ is available on all of the input channels and tape returns. In addition, the TM-D8000 has eight channels of assignable dynamics processing. Snapshot and dynamic automation are available, and there are libraries for the EQ and dynamics processors. Four subgroup faders can control any combination of grouped faders. Like the TM-D4000, the TM-D8000 provides transport controls, complete with a jog wheel. A Mac-based automation package ships with the mixer.

Designed much like a traditional analog console, the Mackie Digital 8-Bus boasts a powerful and flexible mixing environment, with a number of routing and processing options. The D8B has four I/O expansion slots, as well as slots for installing Mackie or third-party DSP boards.

Mackie D8B. The long period that Mackie spent developing the Digital 8-Bus has paid off: it is one of the most intelligently designed digital mixers on the market. Set up similarly to an analog console with numerous dedicated buttons, the D8B packs a lot of power. We're talking high-level features, including 12 aux sends and 16 returns, as well as support for 7.1 surround sound. With its dedicated tape returns and the number of digital formats supported, the D8B is perfectly designed for a 24-track studio of any kind (MDM, analog, DASH, and so on). And, with third-party effects on their way, the D8B moves us another step closer to the concept of the "self-contained" digital mixing environment. If you spend the money on this high-priced console, though, I'd get a monitor, mouse, and keyboard to operate it.

Panasonic/Ramsa WR-DA7. Designed with the musician in mind, the WR-DA7 is user-friendly and intuitive. It has lots of dedicated control sections (including routings, dynamics, and EQ) that make working with the console a breeze. The abundance of I/O options allows you to fit it into any type of studio, although I would prefer to have a few additional XLR inputs on the stock console. The big drawback is that it has no onboard multi-effects processors; considering this limitation, having only six aux sends might be a bit of a problem. Otherwise, though, the WR-DA7 is one of the more powerful mixers discussed here, offering EQ, dynamics, and delay on every channel. It's an excellent console for a 16-track MDM studio, and its MIDI control features are extremely useful in a sequencing environment. However, you need to purchase a few expansion cards before you can do any serious work with this unit.

Roland VM-3100Pro. A brand-new offering from a company that knows how to shake things up in the industry, the VM-3100Pro fills a distinct void in the market, fitting right between the TM-D1000 and the VM04. Although its

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interface is a little confusing, this mixer sounds good, offers solid effects (albeit limited dynamics processing), and delivers some features that you wouldn't expect on a mixer of this size, including MMC control, multichannel digital I/O, and more. It's a nice choice for the owner of one MDM, especially if you're on a tight budget. I would welcome some additional XLR inputs, though.

Spirit by Soundcraft Digital 328. In terms of I/O, the Digital 328 might be the ideal console for a 16-track ADAT or DA-88 setup. It provides XLR and ¼-inch jacks for all of the channel inputs, as well as TDIF and Lightpipe ports to handle the tape returns. There are no I/O expansion options, so you can't have multichannel AES/EBU I/O. Analog junkies will feel right at home with its horizontal channel strip and wealth of dedicated channel meters. The Digital 328 has a great-sounding EQ and some solid multi-effects processing. Keep in mind that dynamics processing is limited to two channels. However, there are inserts available on every input channel (but not on the tape returns).

Tascam TM-D1000. The TM-D1000 is a great choice for a single DA-88 or ADAT setup, provided you have a TDIF-to-Lightpipe converter. (See "Digital Pipelines" in the April 1999 **EM**.) The interface is easy to navigate, the EQ and effects sound good, and there's plenty of digital I/O (both standard and optional). However, even with the effects expansion card installed, dynamics-processing power is still somewhat limited, and you won't be able to put a compressor or gate on every channel. (Then again, do you ever really do that with a DAW?) This is something to keep in mind if you're planning to use the console with a 16-track setup.

Yamaha 01V. Like the TM-D1000, 01V is a fine choice for a single-MDM studio. I especially like the fact that XLR mic inputs are provided for every analog input channel. As with the 02R and 03D, its DSP is fantastic, and the console sounds crystal clear. One drawback to the 01V is that channels 17 through 24 aren't full-service, which means that if you route signals from your multitrack to the console digitally, you have access to only two bands of EQ at mixdown. However, you can swap the tape returns with the input channels, giving you access to the full EQ complement of the input channels.

Yamaha 02R. The 02R is the oldest mixer of the group, yet with all the new offerings on the market, it still holds its own. Though certainly not the easiest board to use, it is one of the most powerful. The 02R is ideal for a 16-track setup (considering its 16 dedicated tape returns) and could also be used in a 24-track or larger studio, for just about any type of production: music, dialog editing, scoring—you name it. It has great-sounding EQ and effects processing, and the 20-bit converters are top-notch. The abundance of I/O options makes the 02R flexible enough to find a home just about anywhere. This console is a veteran, and I don't think it'll be going away any time soon.

Yamaha 03D. Because the 03D was released after the 02R, Yamaha was able to consider feedback from 02R owners when designing this mixer. The result is a slightly more user-friendly console with a number of interface improvements—a particularly important consideration because the 03D has far fewer dedicated knobs and buttons than the 02R. Granted, the 03D's interface is still not nearly as intuitive as that of some mixers discussed here, but all of the refinements—especially the addition of a mouse port—make working with the console relatively painless. Like the 02R, the 03D offers solid EQ and effects processing. With its surround-sound capabilities, the 03D is a smart choice for a midlevel post-production or music-production house or for a sound designer working at home.

EM *Associate Editor* **Jeff Casey** *is probably the only person on the planet who has too many digital mixers in his studio.*

DIGITAL MIXERS IN REVIEW

Although **EM** hasn't yet had the opportunity to review all of the products discussed in this article, we have reviewed many of them. Most of the others are brand-new, and we plan to review them in the near future.

Also, be sure to see the digital-mixer feature "Studio Heart Transplants" in the February 1998 issue. All of these articles are available in the Article Archives section of the **EM** Web site at www.emusician.com.

Panasonic/Ramsa WR-DA7	November 1998
Spirit by Soundcraft Digital 328	March 1999
Tascam TM-D1000	July 1999
Yamaha 01V	October 1998
Yamaha 02R	July 1996
Yamaha 03D	November 1997

A great choice for the single-MDM studio, the Tascam TM-D1000 comes standard with a TDIF port and two multi-effects processors.

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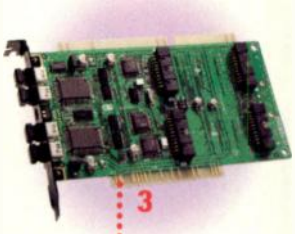

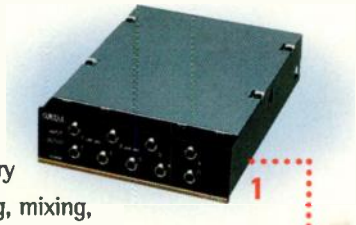

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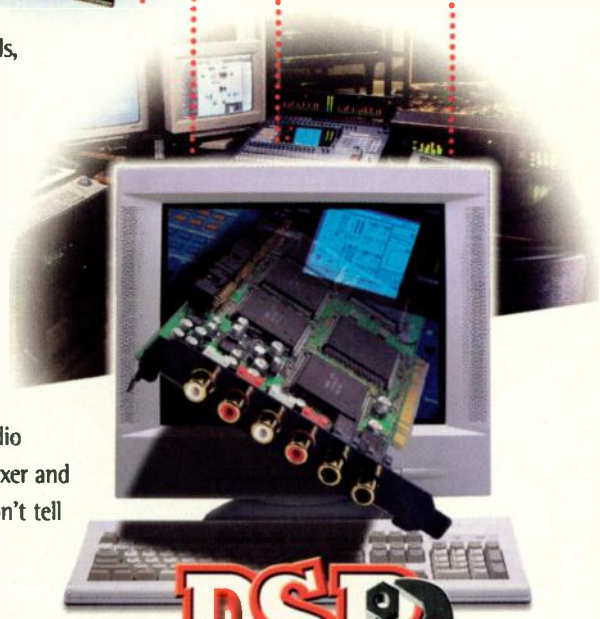

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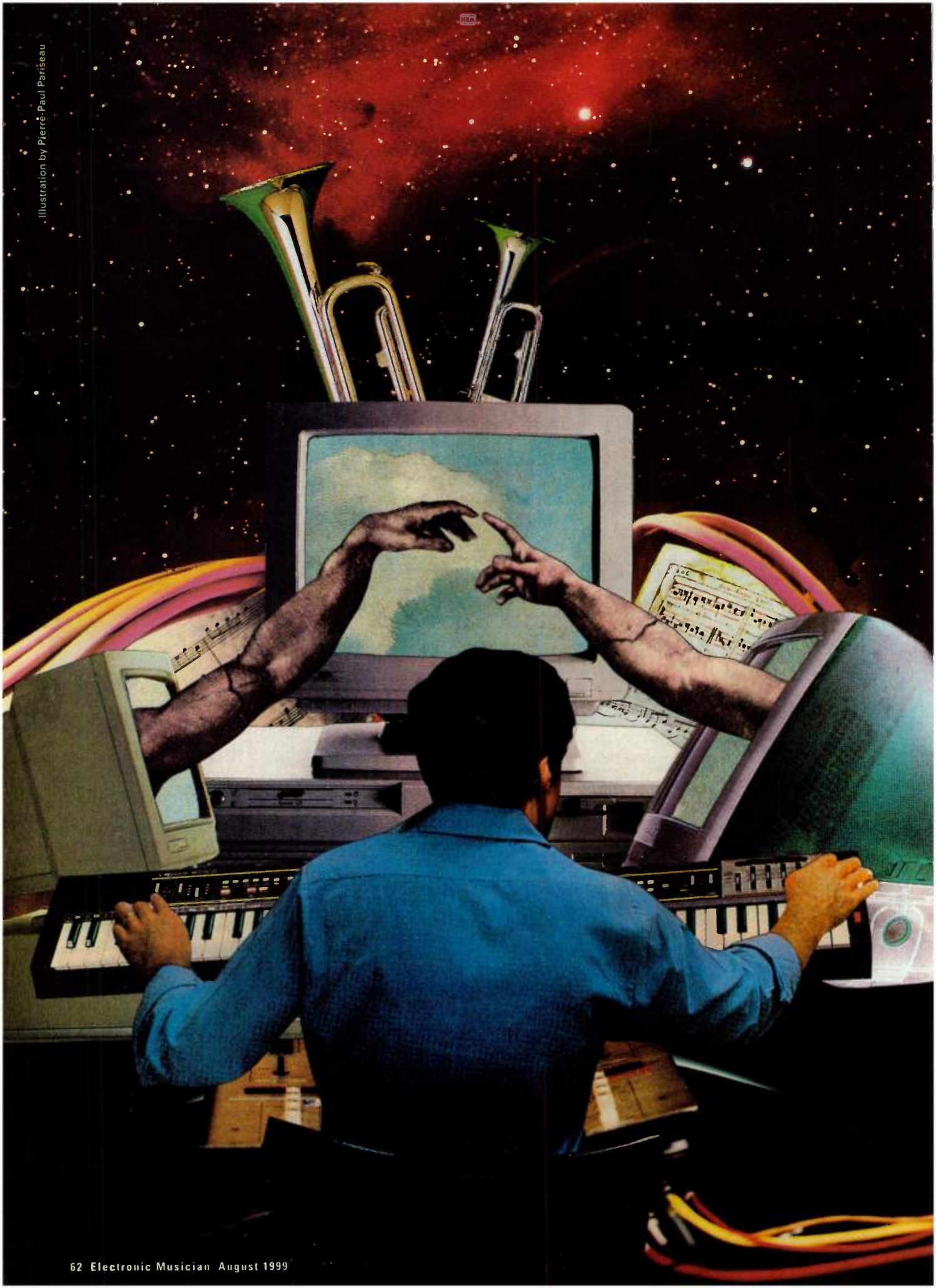

Illustration by Pierre-Paul Pariseau

Linking Up

The hows and whys of networking personal computers.

By R Pickett

In the movie *Back to the Future*, Michael J. Fox's character has traveled back to the 1950s and is asked whether he has a TV. "Sure; we have two TVs," he alleges, prompting disbelief from his audience: "Oh, he's joking—nobody has two TVs." The joke, of course, is that in the 1950s televisions were a luxury that not all households owned, whereas 30 years later, a household with several televisions had become common. In a sense, we are emerging from the "'50s era" of personal computers, as more and more households are finding themselves with two or more computers. Indeed, the vastly increased interest in the Internet over the past several years has led not only to increased computer sales, but also to a growing awareness that by networking computers together, their usefulness can be multiplied exponentially.

The need for a network can stem from many sources. Studio operators usually have a single computer coordinating the tangle of gear and serving as the nerve center of their studios. But innovation marches on, machines become faster, the application landscape changes, and eventually it becomes time to upgrade. For musicians, this is a particularly daunting task: expensive hardware like Pro Tools or old, reliable versions of software might not be compatible with a new machine. They might also be interested in software that runs on another platform but reluctant to abandon the current one. Concerns like these have left many studios limping along on decrepit old computers because their owners want to avoid the problems that an upgrade might bring. Building a network can sidestep these issues and often can result in an integrated system more powerful than its individual components.

Farther up the chain, each local telephone company must agree on how to route the connection, so that your aunt's phone will ring. And finally, you must agree on a topic; nothing is going to get done if you want to talk about politics and your aunt wants to talk about the weather.

Because our objective is to get a small number of machines at the same site connected and able to share files, we're going to work with a very simplified networking model of only three layers, which I'll label Physical, Network, and File System, from bottom to top. We'll consider the pros and cons of various schemes and hopefully come out with an idea of how to set up a small peer-to-peer network.

PHYSICAL LAYER

The Physical layer comprises the type of cabling that connects the networked machines, and the electrical or optical signals that the networking hardware generates. Several schemes have been popular over the years, but most have fallen into disuse. We'll take a look at LocalTalk and Token Ring as two examples, but our focus will be Ethernet, which is by far the cost/performance leader for small networks.

LocalTalk refers to Apple's networking scheme that uses Macintosh serial ports. LocalTalk's best feature is its simplicity: plug machines together with serial cables, and they can communicate. LocalTalk's downsides, however, far outweigh its simplicity. Specifically, LocalTalk runs at 256 kbps, only between four and five times faster than a 56 kbps modem. That's completely unacceptable for audio work. Additionally, LocalTalk's use of the Mac's serial ports can cause disruptions to a MIDI interface attached to the networked Mac. Finally, LocalTalk is not easily available for PC-style hardware; nobody is currently making PC-based LocalTalk hardware, especially because new Macs have Ethernet built in. This means that cross-platform work with LocalTalk is out of the question. In general, anyone with even the smallest budget would be happier with one of the faster, cross-platform alternatives.

Token Ring, developed by IBM, was a popular alternative to Ethernet for a while. Adapters exist for nearly all hardware platforms, and all sorts of Network-layer schemes were updated to run on Token Ring. It is much faster than LocalTalk, with flavors that run at 4 mbps and 16 mbps, or 16 and 64 times the speed of LocalTalk, respectively. The primary downside is cost: Token Ring's design is such that a network has to have a costly hub to connect even two computers, and cards for most machines are more rare and costly than Ethernet cards. Still, companies sometimes update their networks and get rid of old Token Ring hardware for pennies on the dollar, so don't turn up your nose at Token Ring hardware if you chance into it for cheap.

The de facto standard for desktop networking, though, is Ethernet. Born at Xerox in the '70s, Ethernet has become the most-installed Physical-layer scheme in the world. Ethernet is fast, running at either 10 mbps or 100 mbps: 40 and 400 times the speed of LocalTalk. Hubs are sometimes necessary to connect more than two machines, but the design of Ethernet makes these hubs very inexpensive. Ethernet is built into all newer Macs and many newer PCs, but if you own a machine that doesn't have Ethernet built in, PCI and ISA Ethernet cards are available for very little money. NuBus Ethernet hardware is a bit harder to come by, but it's still available from a variety of vendors. One other complicating factor for early Macs with built-in Ethernet is that they require an external adapter to connect to the actual Ethernet cable. These are available from various online and specialty outlets. For almost any on-site networking situation, Ethernet is the scheme of choice.

DOWN TO THE WIRE

Once you've decided on the type of network you want to build, you still face the issue of cabling. Both Ethernet and Token Ring can run on either standard coaxial (coax) cable, like the one delivering your cable TV, or an eight-conductor phonelike cable called Unshielded Twisted Pair (UTP). Naturally, there are trade-offs of cost and flexibility between the two approaches.

Older Ethernet cards have coax interfaces. Coax is cheap and easy to come by, but it requires that all of your machines be connected in a line. If you're connecting only two machines, this is not an issue, but if you connect several computers and the one in the middle has a problem, machines on either side of it will not be able to communicate with each other.

UTP, on the other hand, uses a hub to connect more than two computers in a scheme that places each machine at the tip of a star, so the failure of any single computer will not affect the other machines. It is also the only type of cabling that most new Ethernet hardware supports. UTP is available in different quality categories; the extra expense for the top-shelf Category 5 cable will save you hassles down the line. (Category 5 is also required for 100-mbps Ethernet.) If you plan to link only two machines with UTP, you don't need a hub, but you'll need to purchase a special crossover cable, available at most computer stores. Most installations, however, would do well to install a small hub, as this simplifies the addition of other Ethernet devices such as printers, ISDN routers, and cable modems (see **Fig. 1**).

When Ethernet is run over UTP, the combination is often referred to as *10Base-T* or *100Base-T*, depending on the speed of the Ethernet. This is how

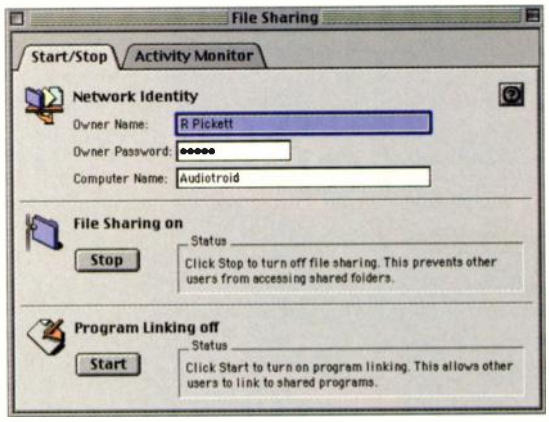

FIG. 2: To run a network on a Mac, you must access either the File Sharing or Sharing Setup control panel on each machine and specify a machine name, user name, and password.

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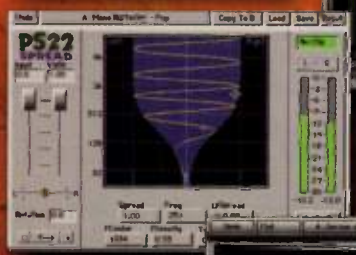

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Ethernet cards and cabling are usually described in catalogs, and it's the way I'll refer to that combination hereafter.

NETWORK LAYER

The lowest-level, software-only layer, the Network layer, deals with issues of encapsulating data into "packets" of information, wrapping those packets in addresses to ensure they get to their destination, and reassembling the data on the far end. This layer is not concerned with the actual information inside the packets; it's just the postal carrier and doesn't read the mail. We're going to take a look at AppleTalk, NetBEUI, and TCP/IP as options for this layer.

AppleTalk is the Macintosh operating system's native networking scheme, and like all things Apple, its claim to fame is ease of use. Plug any AppleTalk device into a network, and it will find all of the others almost immediately. The downside is that this ease of use causes a large performance hit compared with other network schemes. All AppleTalk devices broadcast a large chunk of information about themselves periodically to all other machines on the network. Additionally, AppleTalk has a very small packet size, meaning that the amount of actual data transmitted is diminished by the overhead needed to set up each packet. These two issues make AppleTalk's performance compare poorly with other software schemes at this layer. Still, if only Mac machines are being networked, AppleTalk wins hands-down for ease of implementation.

The analogous Windows protocol is NetBEUI, a protocol developed by Microsoft and IBM in the 1980s. As protocols go, it is lean and mean, favoring silence over chattiness. For instance, whereas AppleTalk devices communicate information about themselves continually, NetBEUI devices do so only if they receive a query. Design decisions like this keep line noise to a minimum. This, combined with NetBEUI's fairly generous packet size, means that NetBEUI is the fastest of the three protocols discussed here. Additionally, NetBEUI works just as simply as AppleTalk for plug-and-play networking. The disadvantage is that NetBEUI is completely Windows specific at this time, so if cross-platform considerations are part of your network plans, NetBEUI is not going to work for you.

The grand old man of Network-layer protocols is TCP/IP (Transmission Control Protocol/Internet Protocol). TCP/IP has been around for almost 30 years and is the Network layer that the Internet uses. It is well designed and fast—only about 10 percent slower than NetBEUI—and ships as a standard part of virtually every operating system today. Unlike AppleTalk and NetBEUI, which are designed specifically for small local networks, TCP/IP scales well from two-machine networks up to networks the size of the Internet itself. The downside of TCP/IP is that this power comes at a cost: setting up a TCP/IP network can be challenging, because each machine has to be configured manually. Once TCP/IP is set up, however, you can connect to the Internet at large, which opens up your possibilities immensely.

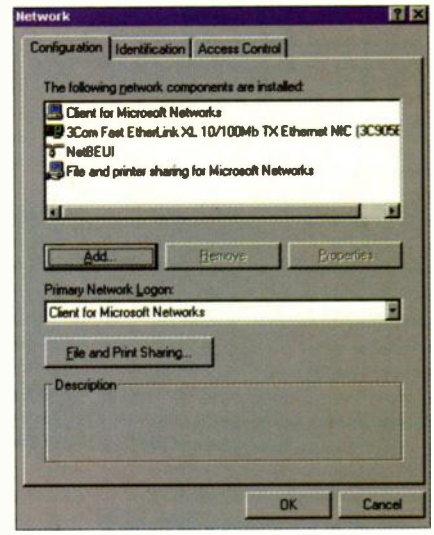

FIG. 3: The Network control panel on a Windows computer contains the network settings that are installed by default. In many cases, you will add options to suit a specific network configuration.

FILE-SYSTEM LAYER

The File-System layer is responsible for actually performing the requests to manipulate files over a network: reading, writing, copying, and such. Naturally, networked file systems use many different schemes. I'll touch on the two major ones for desktop networking, AFP and SMB, with a passing nod to the original such scheme, NFS.

AFP (AppleTalk Filing Protocol) is the file-sharing scheme built into Mac OS. Since System 7, Mac OS has been able to function as both server and client for the AFP protocol, enabling simple and effective peer-to-peer file sharing over AppleTalk networks. As of OS 8, Mac OS has included code that allows AFP transactions over TCP/IP networks, which increases performance dramatically. Currently, this requires a special AppleShare server, but Apple is reportedly working on peer-to-peer AFP file sharing over TCP/IP, possibly as soon as OS 8.6.

SMB (Server Message Block) is the proper name for what Windows calls Microsoft Networking and is actually quite a bit older than Windows itself. It is Windows' native file-sharing scheme but has been implemented in a large variety of operating systems, including DOS, OS/2, and most Unix flavors. SMB runs on top of many Network layer protocols, including NetBEUI and TCP/IP.

NFS is Sun's original Networked File System and is the preferred file-sharing

ALL YOU'LL EVER NEED TO KNOW

A succinct description of the textbook OSI layer model can be found at whatis.com/osi.htm.

For further reading on TCP/IP concepts and configuration, start with "An Introduction to Internet Protocols for Newbies," found at www.halcyon.com/cliffg/uwteach/shared_info/internet_protocols.html. Another good introduction is "Daryl's TCP/IP Primer," available at ipprimer.windsorcs.com.

The ultimate TCP/IP textbook, however, is O'Reilly and Associates' *TCP/IP Network Administration, 2nd Edition*, which, for those with a genuine interest, cannot be recommended enough. You can get in touch with the publisher by accessing www.ora.com.

For manufacturer information, please see Contact Sheet, p. 163.

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mechanism of Unix machines. It is very well engineered and handles error conditions, like the loss of network connection, much more gracefully than AFP or SMB. However, NFS server and client software for Mac OS and Windows is for the most part rare and expensive, so unless you're looking to integrate a couple of desktops into an existing network of SGIs, for instance, NFS is probably not the first choice for this layer.

EXAMPLE NETWORKS

Now let's take all of this theory and put it to use, assembling different home studio networks for various situations. We'll start simply by networking a pair of Macs and a separate pair of Windows boxes just to share files. Then we'll move on to a more complicated cross-platform situation.

Steve MacUser has used his Pro Tools rig on his old Mac IIvx for the past few years, but he recently purchased a new G3 desktop to do sound design and DSP work. His G3 has built-in Ethernet with a 10Base-T port directly on the machine, but the IIvx needs a NuBus Ethernet card. He gets this at his local Mac specialty store or via mail order, and because he plans to network only the two machines, he also buys a Category 5 crossover cable at the same time. Steve installs the NuBus card in the third slot on his IIvx (his Pro Tools system uses the first two slots) and installs the drivers that came with it. He then connects his crossover cable from the 10Base-T port on the IIvx's NuBus card to the 10Base-T port on the back of the G3 and sees by the "link light" indicator on each machine that they are connected correctly. If the light didn't come on, he would need to check his connections, make sure that he is using a crossover cable and not a standard one, and make sure that the cable is good.

Next, because he plans to share files only between Macs, he decides to use AFS and AppleTalk. He configures the AppleTalk control panel on each machine, setting the G3 to use onboard Ethernet and the IIvx to use the card he installed. Then, depending on the Mac OS version he has, he goes to either the File Sharing or Sharing Setup control panel on each machine and sets up a machine name, user name, and password. He then clicks the Start button for file sharing (see **Fig. 2**). To verify that it all works, he opens the Chooser on each machine and clicks on the AppleShare icon to see whether the name of the other machine appears in the list on the right. Steve's network is set up; all that remains is for him to share folders or hard drives from one machine. He'll do this by using the File Sharing menu on the Finder and then mounting the shared items on the other machine using the Chooser and the remote machine's user name and password.

Sue WinUser, on the other hand, has had two Pentium-based Windows 95 machines for some time, situated in different rooms because of space considerations at her studio. One machine drives an array of synth gear that she's been using to create her magnum opus of ambient trance, and the other machine is dedicated primarily to doing live recordings of her Celtic metal band. She wants to use loops and samples of the band in her ambient pieces, so she decides to run an Ethernet connection between them. Because both of her machines have PCI slots, she purchases a pair of PCI 10/100 Ethernet cards and a Category 5 crossover cable. The PCI cards don't need prior configuration, so she installs a card in each machine and then installs the included drivers. Sue connects the two machines with the crossover cable and verifies that the link light is on at each end.

Opening the Network control panel on each machine, Sue examines the list of clients, protocols, adapters, and services. She makes sure that each machine has at the minimum the Client for Microsoft Networks SMB client, the Microsoft NetBEUI protocol, the adapter software that corresponds to the Ethernet cards she installed, and the File and Print Sharing for Microsoft Networks services (see **Fig. 3**). If not, she adds them to the list and, when prompted, inserts her Windows CD-ROM into the drive to install the necessary files. Windows will often install the bulk of these by default, sometimes including other clients and protocols such as the Client for Netware Networks and the IPX/SPX Compatible Protocol. These extra protocols should not cause any problems, but if Sue started to observe strange behavior on her computer, a good first step in troubleshooting would be for her to remove everything except the four listed above.

FIG. 4: Among the newest uses of networking are multiuser applications such as Rocket Networks' Res Rocket system, shown here in action under Steinberg's *Cubase VST*.

Sue also assigns names to the machines and confirms that they are in the same work group by checking the Identification tab of the Network control panel. Now, if everything goes well, she should have an SMB/NetBEUI/Ethernet network set up at 100 mbps (if her cards and drivers support that speed). To verify, she opens the Network Neighborhood icon on each machine and, hopefully, sees an icon for the other machine there. At this point, she can start sharing folders and drives by right-clicking on them and selecting the Sharing menu option.

What if Steve and Sue want to work together on a project? Because both of their platforms are using 10Base-T Ethernet, their networks are compatible at the Physical layer. This becomes more complicated, however, at the higher layers. Let's take a quick look at how to make each platform compatible with the other using third-party software.

CROSS-PLATFORM SOLUTIONS

Miramar Systems (www.miramarsys.com) sells *PCMacLan*, which installs AppleTalk and AFP support for Windows 95, 98, or NT, so they can share files directly with Macintosh machines. I've used *PCMacLan* in several places, and it works simply and perfectly as advertised, with no headaches. At $199 for each Windows 95 or 98 machine and $249 for each Windows NT machine, though, *PCMacLan* is a pricey way to get Windows and Macs talking to each other, especially because it uses the rather poky AppleTalk Networking scheme.

Thursby Systems (www.thursby.com) offers the inverse product, called *DAVE*, which lets Mac OS machines work with SMB-based networks. My experience with a version of *DAVE* earlier than the current one was that it was fast and simple and integrated perfectly into the Mac OS networking paradigm. Current versions continue to get rave reviews. At $149 per copy, it's cheaper than *PCMacLan* but has the mixed blessing of running over TCP/IP. This is good because TCP/IP is much faster than AppleTalk, which makes *DAVE* a better option for timing-sensitive, cross-platform work. However, configuring TCP/IP correctly is a somewhat arcane task that is beyond the scope of this article. (You'll find additional resources listed in the sidebar "All You'll Ever Need to Know.")

On the other hand, configuring a network to use TCP/IP has benefits aside from the ability to use speedy SMB file sharing. A properly configured TCP/IP network can share a single dial-up or dedicated connection to the Internet. This would allow all of your machines access to the outside world by using third-party software or the capabilities built into high-end operating systems such as Linux or Windows NT Server. (Internet service providers that offer this type of multimachine connection usually also offer support for configuring a TCP/IP network to work with it.) Additionally, because TCP/IP can be used for both SMB and future releases of AFS, and it is the network protocol used on the Internet at large, TCP/IP is rapidly becoming synonymous with networking. Accordingly, the sidebar contains sources for information on TCP/IP networking for those so inclined.

GET IT TOGETHER

As the trend toward networking computers continues to grow, more and more music applications are becoming increasingly network aware beyond simple file sharing. For example, the RocketPower technology recently announced by Rocket Networks directly embeds awareness for TCP/IP and the Internet into audio-production software, allowing for collaboration between musicians worldwide, in nearly real time (see **Fig. 4**). Future developments might include being able to assign audio or MIDI outputs to remote machines directly or to distribute intensive signal processing tasks over multiple machines to speed their execution. This could lead to power and performance that we can currently only dream of.

Computers will continue to become cheaper, more powerful, smaller, and more network aware. The eventual ramifications for electronic music production have only begun to be explored. Putting together even the simplest network, however, can increase the value, power, and longevity of your current hardware. And it gives you real-world experience in the art and science of interconnecting computers.

R Pickett *(emerson@hayseed.net) has assembled and administered networks for a variety of people and companies, some of which you might have even heard of.*

Producer/composer Mark Hudson gives us a feel for his production style.

Superstar of the '70s. Songwriter to the stars. Head of Ringo's production team. With a résumé like that, you'd expect someone like Mark Hudson to be unapproachable—hey, most people in this industry are. But not Hudson. He's a genuinely nice guy who, through years of hard work, has created a place for himself in rock's encyclopedia.

Hudson began his show-biz career almost three decades ago as a member of the memorable Hudson Brothers musical troupe. From there he has forged a career as a respected musician, producer, and composer, writing songs for Celine Dion ("Reason") and Aerosmith ("Living on the Edge"), among others.

Recently he has been working with a variety of legendary artists. In fact, in producing Ringo Starr's *Vertical Man*, Hudson's collaborators included George Harrison, Steve Dudas, Joe Walsh, Steve Cropper, Jeff "Skunk" Baxter, Steven Tyler, Paul McCartney, Timothy B. Schmit, Brian Wilson, Alanis Morissette, Scott Weiland, Ozzy Osbourne, and George Martin. Now that's what I call an all-star session.

As a producer, Mark Hudson is known for his reluctance to let technology and money shape the way music is made. *Vertical Man*, for example, was recorded in a makeshift studio set up in a converted office above a Thai restaurant in West Los Angeles. The Ozzy Osbourne record he recently produced was done not with the typical SSL G-Series/Studer A880 combination but actually using a 4-track Tascam Portastudio. Anyone who can convince Ozzy to record his album on a cassette 4-track is a hero in my book. To Hudson, it's all about the vibe, not the gear.

Hudson was gracious enough to spend some time with us talking about his studio philosophy, his thoughts on technology, and his approach to writing and producing music.

By Jeff Casey

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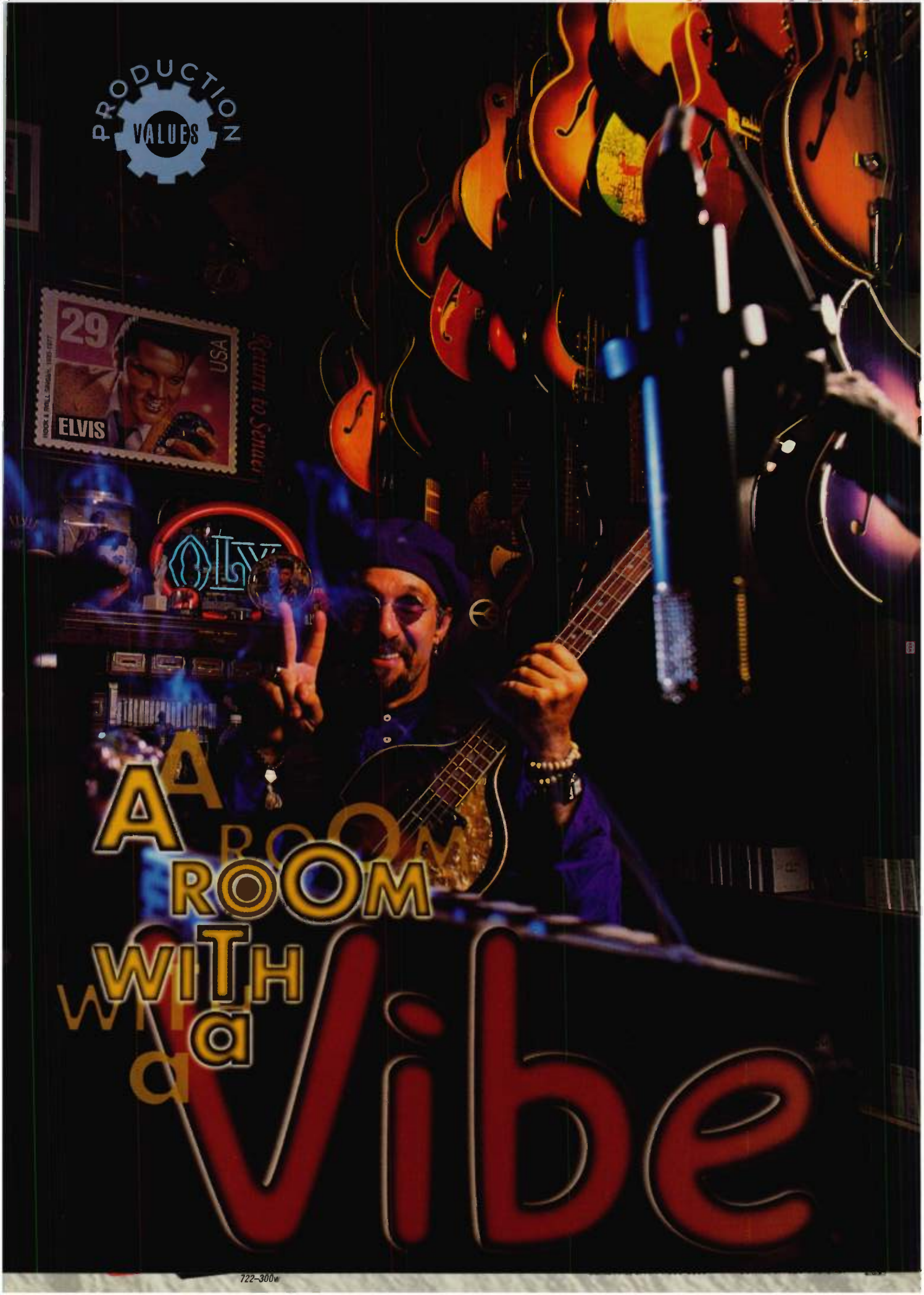
PRODUCTION VALUES
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USA
ELVIS
A ROOM WITH a Vibe

That leads me to my next question: what are your thoughts on the issue of analog versus digital?

There's no question—I'm still an analog guy, although I like the DA-88s. Even when I work digitally, I'm still running stuff through equipment that makes it sound pretty noisy. As [drummer] Jim Keltner said—and I kind of agree: "In digital, you never hear the sound coming and you never hear it going." The signal sort of hits you immediately.

That's an interesting way of putting it. Was it Geoff Emerick that mixed *Vertical Man*?

Yeah. That was a lesson for me. If every engineer in the industry watched Geoff Emerick for a day, they'd either quit or completely change their way of thinking.

For instance, when mixing the Ringo record, he did stuff like turn the high end all the way up and get rid of the bass completely. It sounded crazy. Then he put the bass back in, after compressing it heavily with two Fairchilds, and it became unbelievable. He tweaks the high end first and gives each instrument its own sonic world. I learned a lot from watching him work.

When I talked to Paul [McCartney] about the Beatles, he told me that they would just turn knobs until everything sounded good. It doesn't matter if you can read a chart and know that you're down 4 dB at 9 kHz. What difference does that really make if it sounds beautiful? I'm not trying to rag on technology, though, because we certainly need it.

Technology makes things easier for us.

It does, especially for collaboration between studios. When I can be in London working with a vocal track that Tom Petty recorded in L.A., and it's all going to the same tape, it's a beautiful thing.

But in general, I think that there are a lot of engineers who don't know how to do the simple things, like miking instruments. A lot of production today is based on computers. I was trained by Phil Ramone and learned a lot from him about miking and ambient recording. I haven't seen that style of production since the days I stopped working with him. We've moved away from that, because the technology is too accessible.

Have you been close-miking most of the instruments in your studio?

This is going to sound weird. I just try different mics, and I point a microphone toward the noise. I wish I could say something brilliant, but that's really what goes on. A magazine called me up wanting to know how I got this one great guitar sound on the Ringo record. I told them that the mic cord was 14 feet long, and the amp was 17 feet away, so I just stretched the cord as far as I could and pointed it at the noise. They laughed, but I told them I was serious.

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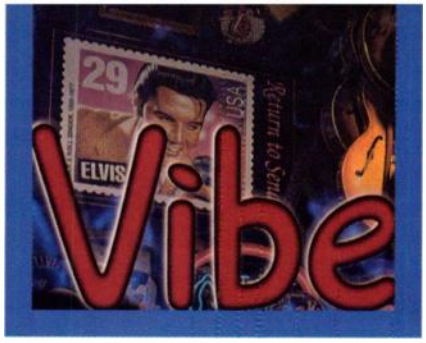


So it was pretty much the guitar signal transmitted through RF?

Oh yeah, and it was great. Now, I'm not really that dangerous—there are miking techniques that I do end up using, especially with a kit like Ringo's. Geoff Emerick suggested that we use a heavily compressed ribbon mic, because often when you do that, you only need one mic on a drum kit.

Overall, I think miking is very important. But I also think that you need to get the sound you want from the instrument, not from the board. I love running through a Neve preamp for the warmth—but still, you have to put your head in front of the speaker and make it sound good. I've also got that John Lennon thing going with the tape slap. I love that effect.

Are you really using tape echo?

Yes, I have two TEAC machines, and I run them like it was done in the old days. And I have this other piece of equipment that I speak very highly of, the Alesis Midiverb III. There's a tape-echo preset in it that is dead-on. We all just say, "Give me more 51," because when you go to preset 51 you become Elvis or John Lennon. You don't have to make the delay time later or earlier, you just pull it up and you're ready to roll. Go to 52 and you're Robert Plant singing "Whole Lotta Love."

I wish we could do away with setting so many parameters on this equipment. I just want to press a button and get the "Whiter Shade of Pale" sort of sound, you know? Because you've really got to know your stuff in order to replicate sounds by tweaking parameters. And as soon as you have to spend time doing that...

For the *Vertical Man* sessions, people were invited in to socialize and even play on the record. One drop-in collaborator was Alanis Morissette, shown here with Ringo and some of Hudson's guitars.

The vibe is lost.

That's right. I could go to lunch and come back by the time the guys figure out how to set the oscillator! It's funny, you can get Midiverbs really cheaply now. I was amazed that people thought I was crazy when I sold my Yamaha DX7 for a B3 organ when everyone was in love with the DX7. What that tells me is, don't throw away your old shoes, because they'll come back in style.

I guess it comes down to personal taste and what you're doing. For instance, I love drum loops, even though I've been saying analog this and analog that. I think that putting a real drummer on top of a loop is heaven. Then you just pick your poison. But to me, there's nothing better than a guy in the room playing. And leakage is cool. You know, we never got into separation until music got screwed up anyway. In the big-band days, you'd set up and make your record.

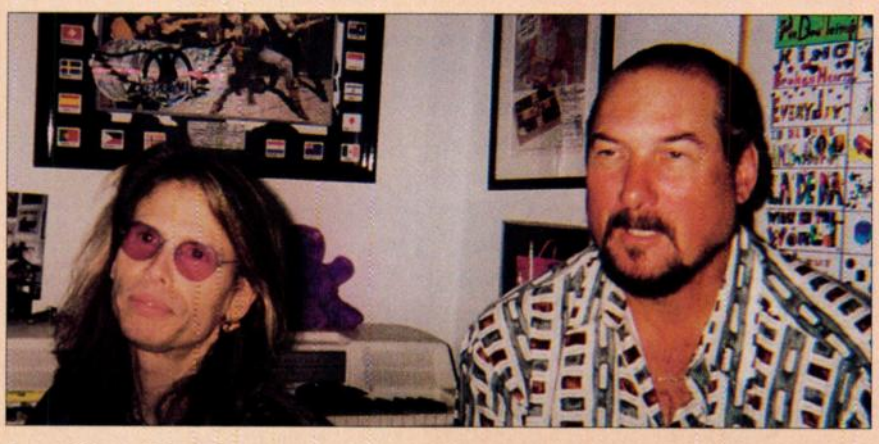
Even the early Beatles records were made like that. I took a trip to Abbey Road recently just to get the vibe of the place. Even though Abbey Road is a big facility, the area where the Beatles recorded is literally the corner of a room—20 feet by 20 feet. So leakage wasn't even an issue then. Once we became concerned about things like isolation, the fun started to go away.

That's why a lot of bands don't like to go into the studio anymore. It's just not fun.

This is where we started our conversation. If I could make my place into a killer studio, that would be ideal. Ringo started this idea that I love. He said, "Anyone who wants to come over can come over. Anyone who can play an instrument will be on the record." So the studio has almost a *Fawlty Towers* quality, where anyone who walks in off the street can hang out. In fact, my landlord wound up playing cello on the title track of *Vertical Man*!

I'm working now with a band called Colony, a new group. As much as I enjoy working with the really successful people, I also love working with the new guys. It's so aggressive and energetic. They can be a lot more innovative. One guy will start singing into a cup, and I'll say, "Why don't you all sing the backgrounds that way?" They're up for stuff like that. I think that the younger generation is finding out that vibe is cool. You have to sound good, but you also have to *feel* good.

On the other hand, as a producer, you have to accommodate the band. That's another thing that we've gotten



Steven Tyler and Steve Cropper enjoy the vibe in Hudson's studio, where a single space serves as tracking room, control room, and rumpus room.

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away from. Producers have become too big, and engineers are often bigger than the producers. We're sitting in an industry where you can spend $300,000 just on the people behind the scenes.

Now, I don't mean that those people don't count. I truly believe that the entire production is a team effort. I just feel that too much credit is given to too many people. I'm only as good as the people I'm working with. However, you do need a "general" to run the show; making a record socialistically just doesn't work. Most of the great records, from the Doors to the Police, had one person who kept it together, even though the other people were part of it. (I always call that person Captain Kirk.)

And, you know, I sometimes knock heads with my engineer, Scott Gordon, because I want to do stuff sonically that breaks the mold. For instance, I want to record strings out in the hallway, because I love the natural ambience. But my hall is only about four feet wide, and there's a huge stairwell, and I want to do a 25-piece orchestra in there. And Scott will say, "Well, how am I gonna mic this?"

One of the songs I wrote for Aerosmith is called "Living on the Edge"—and that's what I do. I would rather be hung from the highest tree for being dangerous than be sitting in a hot tub with 57 other guys who all do everything the same way. I think experimentation is what makes this industry rock. Like when rap first came in, it was a bunch of urban kids' slant on what they thought was cool.

Hudson takes a moment for frivolity with Paul McCartney and Ringo Starr during the sessions for *Vertical Man*, which he calls "basically a Beatles album without John."

I didn't know how to mic Brian Wilson. He came to sing on the *Vertical Man* sessions. But he sings really soft, and I didn't know what to do. He triple-tracks vocals, so when it's built it sounds large. But the C 12 was too sensitive, and you could hear him breathing and everything. So I tried a Shure SM57, but that went too far the other way. I ended up using an AKG C 14; it was crispy and had a cool high end, which was perfect for Brian's voice.

I write all this stuff down because then I can look back and remember how we did things—especially when I was working with Geoff Emerick. He has so many cool tricks, like making a Leslie slow down or miking the orchestra through a toilet-paper roll. I always thought that I was groovy and different, and that guy just took me to lunch!

***Vertical Man* was recorded in Hudson's office, above a Thai restaurant in West L.A.**

So you kept a diary throughout the sessions?

Oh sure. I think it's a good thing to do. I learned that trick from an early record I had—I can't quite remember which one—that said on the album cover what mics had been used and how the producer had miked everything. When I saw that, I thought it was so cool, because now and then, with the body of work that I've done, I often say things like, "I'd love to get a sound like we got when we did those hand claps." I can go back to my notes and see that, oh yeah, we brought some plywood down and we all wore wooden shoes and stamped our feet at the same time that we were clapping our hands.

A diary becomes a stock item in your studio. I think that a lot of producers and engineers should journalize their work, because when you get a sound that you think is magic, that sound could make you famous. Remember, all of those people who wrote the book on sound were just experimenting themselves. God bless them for turning that knob.

What do you think is the best album you've worked on?

Wow. *[Long pause.]* I'll have to split it up as a tie. From the perspective of my career as an artist, it would have to be the Hudson Brothers album *Bafa*. With that album the songs were in place, we started getting outside acceptance from our peers, and we started working with other people like the Beach Boys. Sonically it represents the Hudson Brothers more than anything else that we did—from orchestration to rock 'n' roll, with all of our pop sensibility

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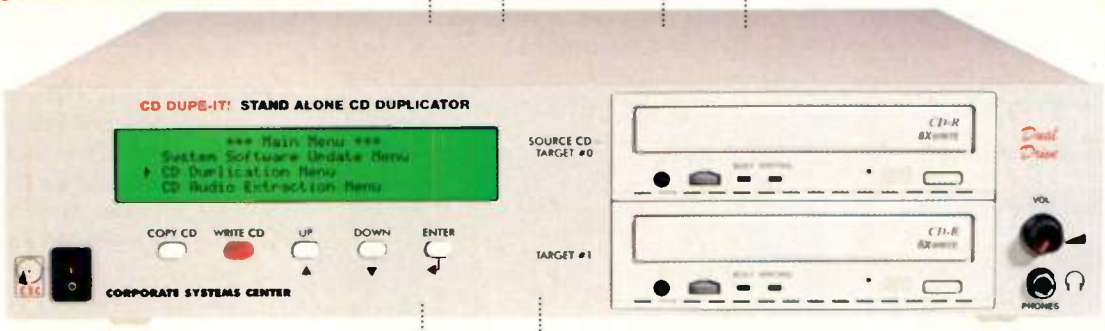

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as well as our Beatles-ish longings.

And, as a producer...I think that *Vertical Man* is, as Geoff Emerick put it, almost like a pack of singles. But to Ringo it was his *Sgt. Pepper*. And Ringo really trusted me—he let me do what I do.

If you could work on any project—as producer, player, arranger, whatever—what would it be?

I would love to produce Paul McCartney. I think—and I don't mean to sound pretentious—that there's a John Lennon element in me. I don't mean talentwise, but I believe I have some of his spirit, which I think Paul is missing: the darker, cynical rock thing.

Right next door to that, I'd like to produce a Steven Tyler solo album. He also has that John Lennon spirit. But the list goes on and on. I'd love to produce Gladys Knight or Madonna; I love all kinds of music. Because of my Beatles influence, I would love to have produced Elvis.

It's truly unfortunate that, as times change, we start changing our thinking. Sometimes if it ain't broke, don't fix it. That doesn't mean that you can't have a contemporary drum sound or whatever. But some of these bands come back and try to reconstruct their world so far away from the things we loved, and it turns the audience off.

When I did this Ringo record, which was basically a Beatles album without John, everyone rocked. The older guys were just as hip as the young guys, and the younger guys were just as groovy as the old guys.

I keep talking about emotion, because I think emotion translates into sound. When you hear "Good Lovin'" by the Young Rascals, you know they were in a room kicking ass. And that, to me, comes out sonically. When it becomes too much of a business, we're in trouble. I do it for free. Yeah, I love making a living—but at the same time, I still do it for nothing. If I go to my studio today and I don't have a job, I'll still be doing the same thing. That's the love of the art. ●

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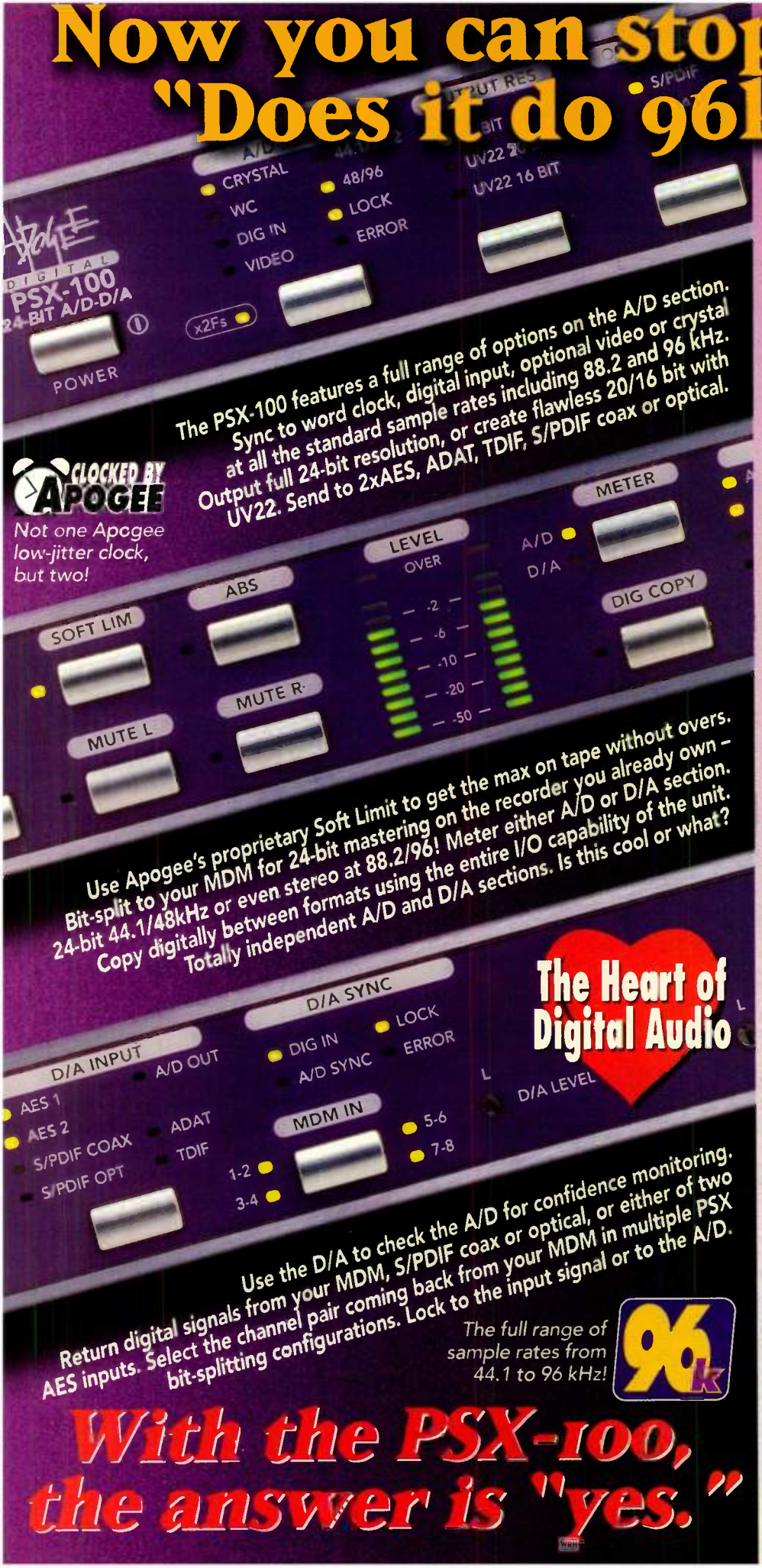

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SQUARE ONE

Play Your Cards Right

Know what to look for when shopping for your next sound card.

By Scott R. Garrigus

When you buy a complete computer system these days, it usually includes audio capabilities. Windows computers come with a basic sound card or audio circuitry on the motherboard, and Macs have built-in audio. Many people settle for the sound already installed on their newly purchased computer system; but what if you want something better? With so many sound cards on the market and so many features to choose from, how do you find a suitable solution to your sonic needs?

DAVID BALL

In this article, I'll discuss the different types of sound cards and explain some of their main features. I'll also talk about basic specifications, what they mean, and what you should look for when buying a new card.

WHAT'S YOUR TYPE?

Sound cards are available in a variety of configurations. The most common is the digital audio/synthesizer sound card, which consists of a built-in MIDI synthesizer and basic digital audio recording and playback capabilities. These are typically consumer-level cards, which can be used for everyday computer tasks such as playing games, running multimedia CD-ROMs, and playing audio and music files from the Internet. This type of card can also be used for creating music and sound as a hobby. Two examples are the Turtle Beach Montego A3DXstream (see **Fig. 1**) and the Creative Labs Sound Blaster Live Value.

The sampling sound card is the next step up, and depending on the amount of RAM available, it can be used in both hobby and professional music-making situations. Unlike the basic sound card, which provides only a predetermined set of sounds, the sampling sound card can be used to create your own personalized sound set by allowing you to record short segments (samples) of audio and store them in the card's memory. (You can also find hundreds

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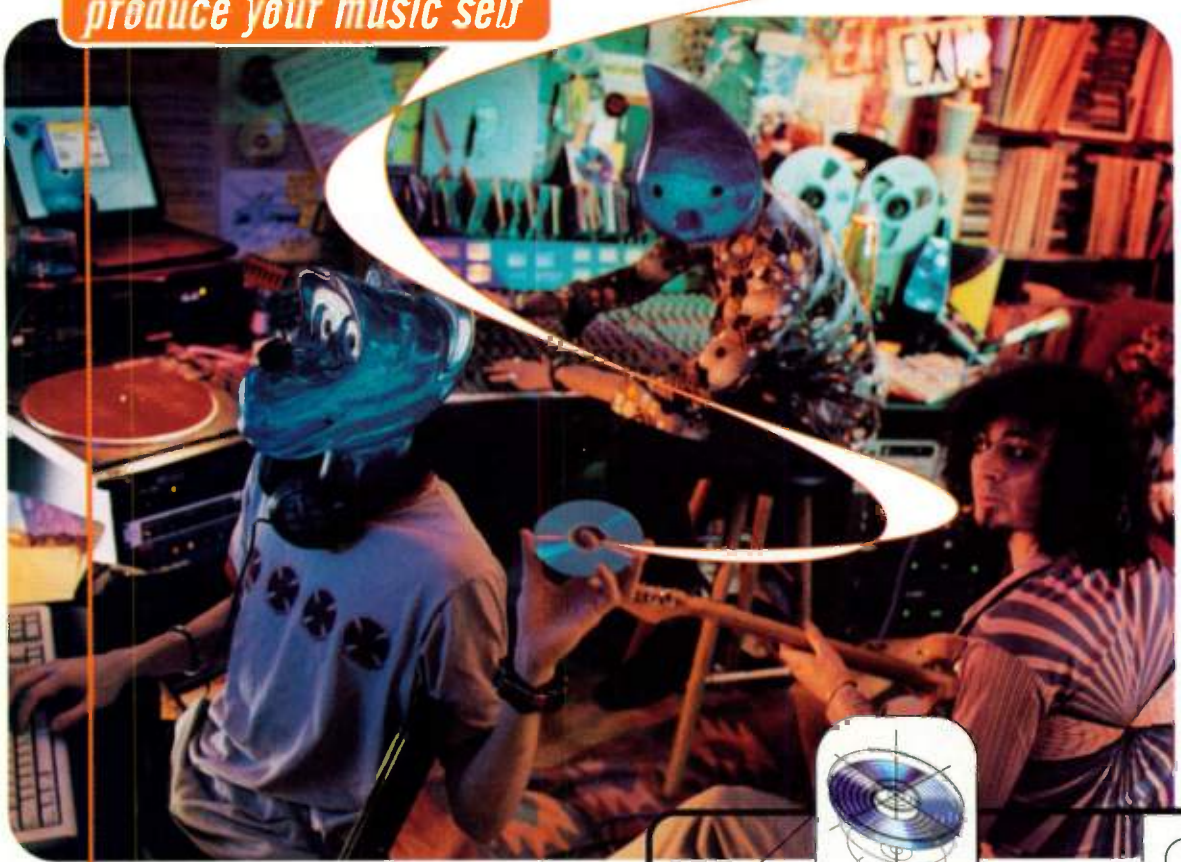

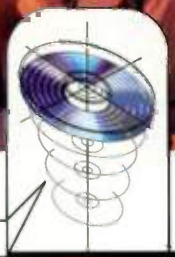

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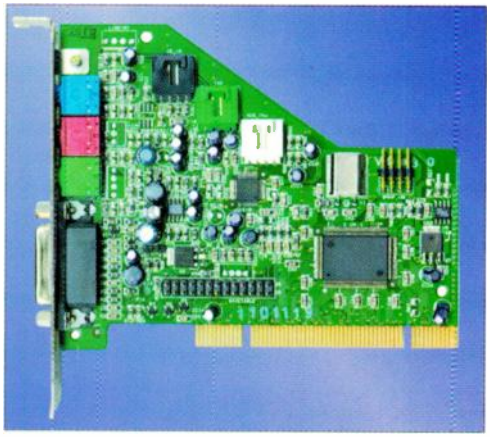

FIG. 1: The Turtle Beach Montego A3DXstream is a good entry-level sound card that is often bundled with new computers. This PCI card provides a 64-voice wavetable synthesizer and full-duplex operation.

of megabytes of samples on the Internet, which can be used for the same purpose.) These samples can then be edited and later played back via MIDI. The E-mu APS card and TerraTec EWS64 series fit into this category.

Finally, the audio interface card is used for recording and playing digital audio and typically for nothing else. Audio interface cards are mid- to high-end sound cards used for recording professional-quality digital audio onto the computer's hard drive. Among the most common uses for this type of card is multitrack recording, in which a digital audio sequencer or multitrack audio program is used to record several tracks of digital audio, sometimes simultaneously. This material can then be mastered and burned onto a CD using special software. The Darla and Gina cards from Event Electronics belong in this category, as do the Frontier Designs Dakota (see **Fig. 2**) and Antex Studio Card AV, both of which add MIDI capabilities to their multichannel recording functions.

PLAYING THE SLOTS

Computers have one of two types of slots for installing cards: ISA (Industry Standard Architecture), which is older and slower; and PCI (Peripheral Component Interconnect), which is newer and faster. (A third type of slot, PCMCIA, for Personal Computer Memory Card International Association, is usually found only on portable computers; sound cards in this format are still rare.) All other things being equal, a PCI card can transfer data faster than an ISA card and, therefore, provides better performance. If you have a Mac, PCI is your only choice.

PCI cards can be more expensive than ISA cards, but you won't find a huge price difference nowadays. Sound card prices range from around $20 to more than $1,000, and, as with most things, you get what you pay for. For under $100, you can get a basic digital audio/synthesizer sound card that is fine for everyday audio needs. Move into the $200 range and you'll find a number of nice sampling sound cards. Climb to $500 or more, and you'll have a professional digital audio recording interface that you can use for top-notch music production.

DIGITAL AUDIO SPECS

The typical sound card provides a number of audio inputs and outputs, including line level, microphone level, and speaker (see **Fig. 3**). Line level inputs and outputs are used to transfer sound from a cassette deck, radio, electronic keyboard, or any other standard audio device. Microphones generate a very low level audio signal, so they need a special input, which is connected to an internal preamplifier on the sound card. Speakers also need their own connector with a built-in amplifier in order to produce a reasonable level of volume. However, you'll probably want to get a good set of powered speakers (or an external amplifier and studio monitors) and then plug them into the line-level output—you'll get much better sound that way.

Some high-end sound cards also offer digital inputs and outputs. These special connectors let you attach the sound card directly to compatible devices such as DAT (digital audio tape) decks and certain CD players. Using these connections gives you the best possible recording quality because audio signals stay in the digital domain without needing to be converted into analog signals. You can also expect to find connectors for Alesis ADAT and Tascam TDIF devices (called modular digital multitracks, or MDMs) on many professional cards. These allow you to transfer 8 or even 16 tracks of digital

RESOUNDING NET RESOURCES

A good place to find additional information before shopping for a sound card is the Internet. You can look at manufacturers' sites for specific product features or peruse newsgroups for answers to many of the questions that you might have. You can also find a number of sites dedicated to music hardware. Once you buy a card, be sure to get hold of the most up-to-date drivers for your card. This can help you avoid problems down the line.

Newsgroups
rec.audio.pro
rec.audio.marketplace
rec.audio.opinion
rec.audio.tech
comp.sys.ibm.pc.soundcard.music
comp.sys.ibm.pc.soundcard.tech

Music Hardware and Software
www.servtech.com/public/koberlan/zmanufac.htm
One of the most complete lists of music hardware and software manufacturers' Web sites.

Harmony Central Classifieds
www.harmony-central.com/Services/#cla
Instead of buying new, you might want to save a few bucks and check out some of the used sound cards for sale.

Harmony Central Computer Music Resources
www.harmony-central.com/Computer/#cards
This page contains links to Web sites that contain very useful information about sound cards and their capabilities.

For manufacturer information, please see Contact Sheet, p. 163.

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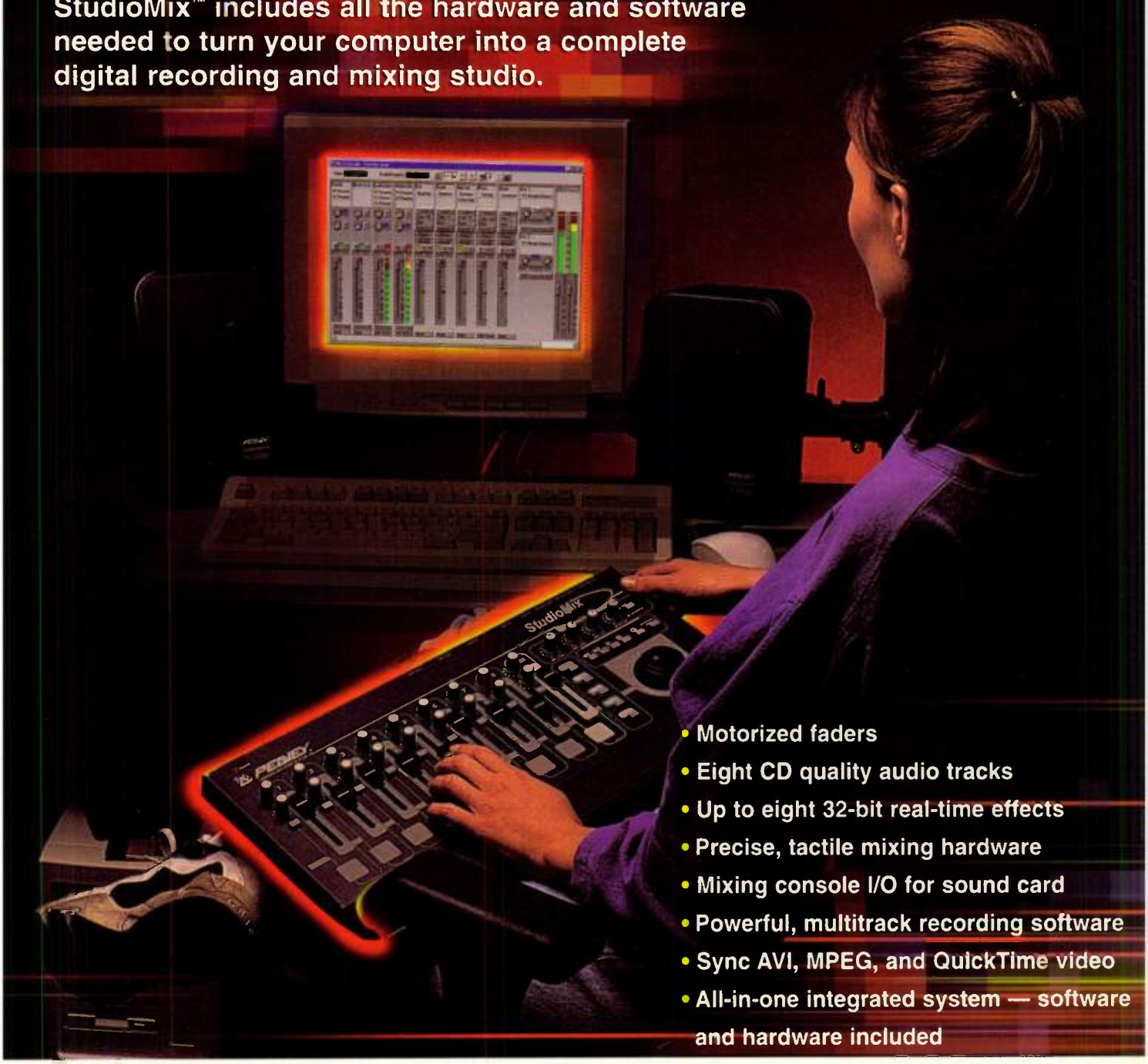

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audio to and from external multitrack recorders.

You should also be aware that connectors come in a variety of forms. Low-cost cards usually provide the same ⅛-inch jacks used for headphones on boom boxes and Walkman-style cassette recorders. For better quality, some cards use ¼-inch, RCA, or XLR jacks. Connections can also be balanced or unbalanced. Balanced cables provide better protection against radio-frequency interference and are more resistant to hum than unbalanced cables.

Even if you can plug multiple devices into a sound card, there's no guarantee that you'll be able to record them simultaneously on separate tracks. Most inexpensive sound cards internally mix all of their audio sources down to one stereo signal. Higher-end cards let you record each source separately on its own discrete stereo channel (assuming your software provides that option), which is desirable in a music recording studio. You should also be aware that not all cards can record and play back audio simultaneously. To do so, the card must have full-duplex capabilities, as opposed to half-duplex.

The quality of the audio signal is, of course, another important consideration. When an analog signal is converted to digital, the sampling rate plays a big part in determining the resultant quality. In general, the higher the sampling rate, the better the sound quality. The sampling rate of a CD is 44.1 kHz (44,100 samples per second), and all modern sound cards support this rate. Professional cards can reach sampling rates of 48 kHz or higher.

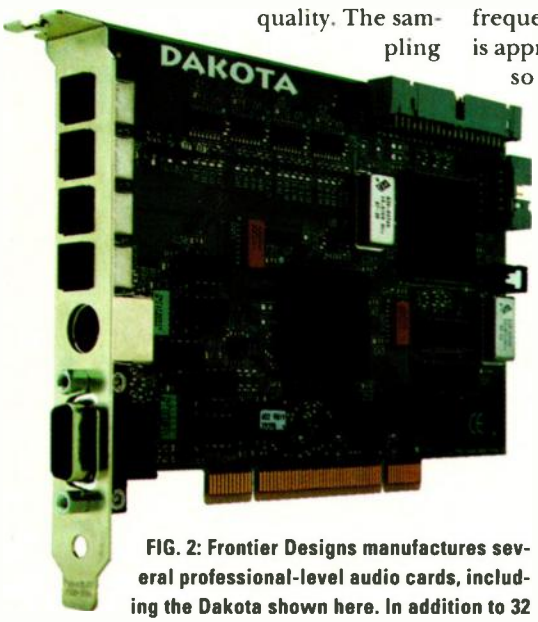

FIG. 2: Frontier Designs manufactures several professional-level audio cards, including the Dakota shown here. In addition to 32 channels of MIDI I/O, the card offers ADAT Lightpipe and S/PDIF digital connectors.

Resolution (or bit depth) is also a factor in determining sound quality. It is similar to the dots-per-inch measurement used in printing: the more dots you have to represent your picture, the clearer it will look. The same goes for digital audio resolution; the more bits you have to represent your signal, the better it sounds. A 16-bit resolution (which provides 65,536 values for measuring a sample) is the CD standard and is supported by all sound cards. Some cards (again, mostly high-end) support 20- or even 24-bit recording.

Two other measurements that you need to consider are signal-to-noise ratio and frequency response. Because all electronic devices produce some noise, the signal-to-noise ratio of a sound card tells you how much greater the signal strength is compared to the amount of internal noise made by the sound card. The bigger the number, the quieter the card. A good signal-to-noise measurement is about 90 dB or higher. Frequency response represents the range of frequencies the sound card can produce (within certain tolerances). The frequency sensitivity of human hearing is approximately from 20 Hz to 20 kHz, so a good sound card should encompass at least that range, if not more.

One final consideration is the inclusion of real-time effects. Digital signal processing (DSP) effects can add a professional quality to any sound mix. Among the most common types of effects are chorus, delay, and reverb. There are different ways to produce these effects, but it's best if the card has a dedicated DSP chip, instead of relying on the host computer to do the job.

SYNTHESIZER SPECS

If you want to purchase a sound card with a built-in MIDI synthesizer or sampler, there are a number of additional features you should know about. Early sound cards incorporated synthesizers based on frequency modulation (FM) synthesis with a limited number of operators. Unfortunately, this method wasn't much of an improvement over the basic beeps and boops of a PC speaker. These stripped-down FM synths work well for organ and bell sounds but fail miserably when trying to portray other types of instruments.

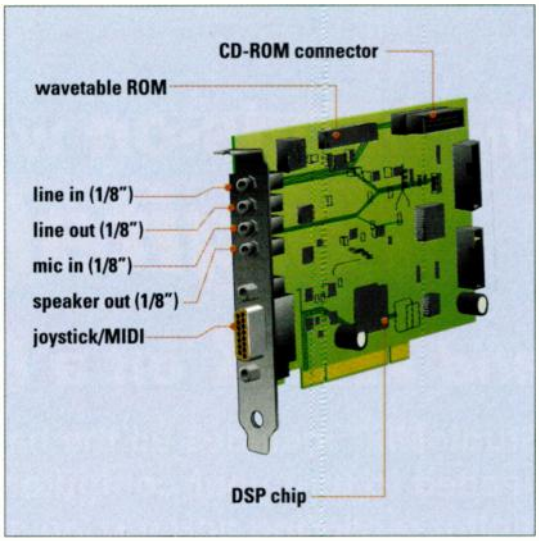

FIG. 3: Sound cards provide connectors for many different devices. Among these are options for MIDI, joysticks, speakers, headphones, and line-level devices.

Today's sound cards use a technology called wavetable synthesis. This process can provide more realistic sounds because the synthesizer plays actual digital recordings (samples) of live instruments. The only drawback to wavetable synthesis is that the samples need to be kept small, because they are stored in RAM or ROM. (Some new cards use your computer's system RAM for storage.) The more memory your card has, the wider the range of sounds it can produce. Look for a bare minimum of 2 MB, but if you can afford it, get a card with 8 MB or more.

Also take into account the degree of playback control over the wavetable samples that a card provides. All sound cards today support General MIDI (GM), which provides consistent numbering for the sounds in your card but doesn't guarantee support for any kind of sound parameter programming directly. That's where the GS and XG formats come in. The Roland GS format expands on the GM standard

by offering additional sounds along with sound-programming control over a few synthesis and effects parameters. Yamaha's XG format goes even further by requiring three separate effects processors and more than a dozen programmable synthesis parameters (such as the brightness of a sound). Be sure to get a card that supports either GS, XG, or both.

BUNDLED UP

In addition to a variety of hardware features, many sound cards also come bundled with numerous types of applications. These range from fairly useless utilities like telephone dialers to respectable, though often scaled back, versions of professional music programs. (Many professional sound cards come without software, however, because the manufacturer assumes that you'll have your own professional tools on hand.) Among the most common applications that you'll find are audio editors, MIDI sequencers, and mixers. These are typically third-party programs that allow you to get started making music right away.

Look for bundled software that allows you to upgrade to a more professional version of the program. *Sound Forge XP* from Sonic Foundry and Syntrillium's *Cool Edit 96* are two PC audio editors that offer this option, as does BIAS's *Peak LE* on the Mac. Common MIDI sequencers include Cakewalk's *Cakewalk Home Studio* and Steinberg's *Cubasis A/V*, both of which also provide an upgrade option. The mixer software that comes with your card is often a no-name utility that cannot be upgraded.

SOUND DECISION

You should now have a good understanding of the numerous features that sound cards provide, and hopefully you're a bit more knowledgeable about what to look for when buying a new one. But before you run off to the store with cash in hand, be sure to make a list of the features that you know you want. It also wouldn't hurt to arm yourself with a little more information about specific cards by visiting a few sites on the Web (see the sidebar "Resounding Net Resources"). If you know what you want before you go looking for it, you shouldn't have any problem finding a sound card that fits your musical needs.

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RECORDING MUSICIAN

Accommodating the Muse

Strategies for recording acoustic guitar and vocals simultaneously.

By Brian Knave

The easiest way to lay down pristine vocal and acoustic guitar tracks is to record them separately—typically, the guitar first and the vocals afterward. The problem with that approach is that it can shortchange the performance, failing to capture the flow, spontaneity, and emotive coherence of a "live" take. Songwriting, after all, is an organic act, usually done with instrument in hand. Overdubbing, by separating singer from instrument, risks also separating singer from muse.

There are technical impositions, too. Many songwriters can't play and sing at their best when they try to separate the two elements. Their vocal phrasing can suffer, or they may tend to lose the song form. Even experienced session players who can readily perform their parts separately will often prefer to sing and play at the same time, particularly when it comes to recording their own material. It just feels better that way. (Another technical advantage of the "undivided" approach is that the singer/guitarist is not required to wear headphones while recording.)

Whatever the case, it behooves the recordist to know how to record vocal and acoustic guitar tracks simultaneously. But be warned: getting a great sound this way is not easy. In fact, it's one of the more challenging applications a recordist can face.

Here are some of my techniques for recording vocals and guitar simultaneously, as well as insights gleaned from Christian Jones, former house engineer at Mobius Music in San Francisco. I got wind of Jones from his work on folk artist Lori B.'s 1998 release *Hurricane Child* (available at www.songs.com/lorib or by calling 1-800-BUY-MY-CD). This heartfelt and well-recorded CD has garnered acclaim from numerous fronts

BRIAN KNAVE

The key to separation when recording guitar and vocals simultaneously is mic positioning—basically, angling the guitar mics down and the vocal mic up. Here, a pair of Neumann KM 184s captures a stereo image of the guitar while a Microtech Gefell MT 711S records the vocals of Chilean singer/songwriter/guitarist Rafael Manriquez.

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and led to Lori B.'s opening for headliners such as Jackson Browne, David Lindley, Bonnie Raitt, and the Jorma Kaukonen Trio. Although *Hurricane Child* contains the occasional overdub, all the basic (guitar and lead vocal) tracks were recorded live in the studio.

ONE, TWO, THREE

The first thing to figure out with the simultaneous approach is how many mics to use. Although it's possible to record decent-sounding guitar and vocal tracks at the same time using a single microphone, it is hardly advisable, because the two source sounds cannot be controlled separately. I recommend using a minimum of two mics—one for the guitar and one for the voice.

You can usually obtain the best results, though, by using at least three microphones: one for the vocal and a stereo pair for the guitar. An acoustic guitar projects different sounds from different areas of its body, and two mics are almost always better than one at capturing this range of tones. Also, a stereo-miked guitar can be panned to fill up more of the soundstage, which is helpful in a sparse mix.

In addition to three close mics, some engineers will also set up a second stereo pair (typically omnis) positioned several feet back to capture ambient sound. Jones, for example, used a stereo pair of Bruel & Kjaer 4006 omnis as room mics. (During mixdown, he brought in a bit of the ambient tracks on certain songs; mostly, however, he used the pair as sends for the reverb, to create a more natural-sounding effect.)

In this article we'll focus on the three-mic approach (stereo guitar, mono vocals). This is the hard part of the equation; once you get the right three mics properly positioned, setting up a room mic or two for added ambience is a relatively simple matter.

LEAKY STYLEY

The main problem that you'll encounter with three live microphones positioned around a guitarist/singer is mic leakage—in this case, vocal sound bleeding into the guitar mics and guitar sound bleeding into the vocal mic. Although you should minimize this leakage as much as possible, you won't get complete separation of the vocal and guitar mics. But that's okay. A little leakage won't ruin the final product; in fact, kept to a minimum, the leakage helps impart a live quality to the tracks.

One way to reduce leakage mechanically is to position some type of baffle (I've seen people use cardboard, felt board, and Plexiglas) between the vocal and guitar mics. A baffle, however, can color the sound with unwanted reflections, so I generally think of it as a last resort. The better approach is to minimize leakage by using correct microphone selection and placement. When done well, this provides sufficient separation and the most natural sound.

MIC SELECTION

Each microphone should have a cardioid, supercardioid, or hypercardioid polar pattern, to afford maximum rejection. Condenser mics are usually best, but you can substitute a dynamic or two if you don't have enough condensers to go around. Most important, select mic types that are flattering to the source sounds.

There's really only one way to determine which mics are best for which sources: set up a variety of mics, record each setup, and listen back through the

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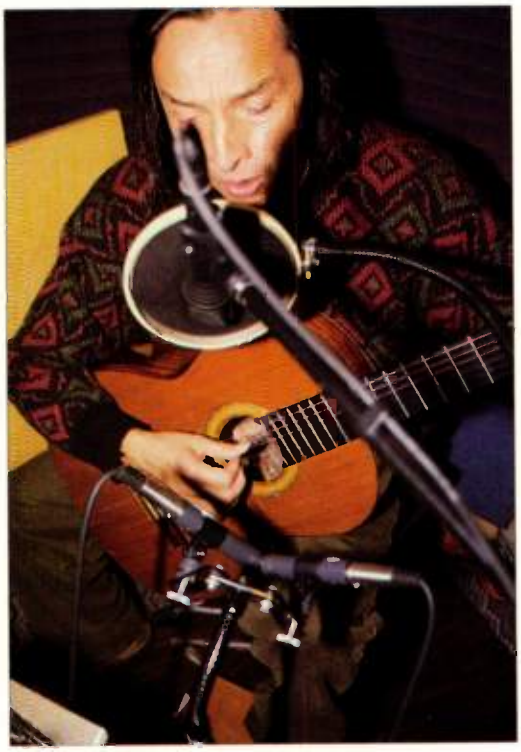

BRIAN KNAVE

FIG. 1: This close-up of Manriquez clearly shows the steep angling of the microphones. All three mics are cardioid, and the polar patterns alone are responsible for creating separation of the guitar and vocal tracks.

monitors to compare. In fact, if you're going for the best sound possible—and have a variety of gear at your disposal—you should also compare mic preamps, compressors, and any other links in the signal chain. (For specifics on comparison testing, see "Recording Musician: Gearing Up for Critical Vocals" in the November 1998 **EM**.)

Such testing, of course, is tedious and time consuming. If possible, perform the mic tests in advance so that everything will be set up and ready to go on the day of the recording session. After doing a few of these tests, you'll develop a sense of the sonic character of each microphone, making subsequent setups go more quickly.

MIC PLACEMENT

Now comes the critical part: positioning the three mics. Again, the goal is to get as much separation as possible between the vocal mic and guitar mics, while also managing to capture a great sound from each instrument. The basic trick is to aim the guitar mics down and the vocal mic up, so that the mics' polar patterns naturally reject all but the desired sound source.

I start with the guitar, which typically is the more difficult instrument to mic. (For more information on miking acoustic guitars, see "Recording Musician: Glorious Acoustic Guitars" in the February 1997 **EM**.) Of the two most common stereo-miking techniques for acoustic guitar—XY coincident and spaced pair—the XY with matched mics is generally best (and easiest) for this application. Not only does it eliminate phase problems between the two guitar mics, but the coincident capsules also ensure that leakage from the vocals hits them at the same time and level—an important advantage, as we'll see.

After setting levels and roughly positioning the XY mic pair in the sweet spot area (which you've predetermined in advance by listening with your ears), have the musician sing and play simultaneously while you monitor with headphones. The trick now is to find a position for the mics that rejects most of the vocal but still sounds great on the guitar.

During a recording session of the brilliant Chilean singer/songwriter/guitarist Rafael Manriquez, I used a pair of Neumann KM 184 microphones on the artist's classical guitar. I found that by positioning the mics nearly vertically to the guitar, pointing toward the floor (see **Fig. 1**), I could still capture an excellent guitar sound. (In this case, it helped that I'd first put down a wood "floor" beneath Manriquez, to create some early reflections from the guitar. It also didn't hurt that the guitar—a hand-built Contreras from Spain—sounded amazing to begin with.) Of course, every mic pair is different, so you have to listen closely to find the best position. Learn to ignore how your setup *looks*. In fact, close your eyes while finalizing the microphones' positions, so you won't be distracted by visual cues.

When it comes to stereo miking an acoustic guitar, I actually prefer the sound of a spaced pair because it tends to capture a greater variety of tones from the guitar. In fact, I often use mismatched mics intentionally, to further extend the tonal variety. Recently, for example, while recording tracks for

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BRIAN KNAVE

FIG. 2: Even though a spaced pair sounds great on guitar, vocals tend to bleed unevenly onto the two mics. Here, guitar wizard Buddy Craig is recorded with a Microtech Gefell MT 711S and an Earthworks SR77 on his guitar and a Shure KSM32 on his vocals. (For the takes we kept, the SR77 was positioned closer to the guitar.)

singer/songwriter/guitarist Buddy Craig, I used two considerably different mics on his Takamine Santa Fe guitar: a Microtech Gefell MT 711S, positioned facing down on the 12th-fret area, and an Earthworks SR77 aimed at the instrument body (see **Fig. 2**). The 711S captured a very contained and nicely isolated sound with the preponderance of low end, along with detail from the strings; the SR77, positioned a bit farther back, provided a brighter and more open "room" sound. Together, the two mics presented a big, full-bodied, and quite realistic portrait of the guitar.

Once the vocal was introduced, however, this setup didn't work as well as I had hoped, mostly because the guitar mics picked up varying amounts of the voice. That's one of the problems inherent to spaced-pair miking (especially with mismatched mics), and it is particularly troublesome in this three-mic application: when you try to mix the three tracks, uneven voice pickup on the two guitar mics—and too much vocal on either or both of them—usually leads to a hollow sound.

Despite the far mic's wonderfully airy and very natural image of Craig's guitar, we had to bring the mic in much closer to the instrument's body to get usable tracks. The problem then, of course, is that the proximity effect starts to happen, and that can result in an overemphasis on bass frequencies. We kicked in a low-cut (75 Hz) filter on the mic preamp, helping reduce the troublesome lows. Largely, though, success with a spaced pair in this three-mic application is hard won and requires careful balancing of the mics and their respective properties.

Jones expressed his preference, too, for a spaced pair on acoustic guitar, to capture "a really broad field." For miking guitar in this application, though, he typically goes with an XY coincident pair—which is what he used for Lori B.'s record. "Lori plays very quietly, with her fingers, so I had to put the mics in pretty tight," Jones explains. "They were about six inches away from the guitar, aimed between the sound hole and the bridge, on a lateral plane, and tilted down to keep out the vocal sound."

DIDN'T PHASE ME

Phase cancellation is another problem inherent to putting up three mics in close proximity. Again, careful positioning is the solution. In this case, however, you use your eyes to make sure the mics conform to the *3-to-1 rule*, which states that the distance between mics should be at least three times the mic-to-source distance. For example, if you position a spaced pair on the guitar with the mics 18 inches apart, each mic should then be no more than six inches away from the guitar. (For more information on this topic, see "Recording Musician: Avoiding Phase Cancellation" in the July 1997 **EM**.)

Fortunately, the 3-to-1 rule is fairly easy to observe in the three-mic application, because the microphones are usually positioned close to the sources to minimize leakage. Just the same, though, sum the three tracks to mono to make sure that no phase anomalies occur. (If your mixer has no mono-sum button, you can mono the tracks simply by panning them all to the same location.)

VOICE FROM ABOVE

Compared with finding the optimal placement for the guitar mics, positioning the vocal mic is a breeze. Just put it at about chin level to the singer, with the capsule tilted up toward his or her mouth, and angled so that the mic's cardioid pattern effectively blocks out most of the guitar sound. Bring the mic in close—two to six inches from the singer's mouth—and by all means use a pop filter.

The amount of rejection you achieve will depend largely on the tightness of the microphone's polar pattern. The Shure KSM32 mic I used on Craig's vocal was amazing in this regard: when I soloed the vocal track, I could hardly hear the guitar at all.

Another effective approach is to use a figure-8 pattern and aim one of the null points (where rejection is strongest—in this case, on either side of the pattern) directly at the guitar and the other at the ceiling. The figure-8 works well in this application due to its exceptionally good rejection at the null points. Of course, if you use a figure-8 you must also make sure that no unwanted sounds or reflections are occurring on the backside of the mic, as these, too, will be picked up.

Once you've positioned the vocal mic, it's time to listen to all three microphones together. Set the fader levels approximately to where they'll be in the final mix and evaluate the overall sound quality. Is the vocal clear and strong, or does it have a hollow, roomy sound? Does the guitar sound natural? Is there enough separation to let you apply different effects to the tracks (for example, chorus on guitar and delay


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XP-30
TONE SWITCH

on the vocal) without ending up with mush? At this point, it may be necessary to tweak mic positions a bit more, to correct any problems and improve the overall sound. Also, remember to sum all the mics to mono to listen for phase anomalies.

In the end, no matter how many mics are in your setup, the sound should be regarded as a single entity—which is what it will be in the final mix. "It's an organic thing," explains Jones. "All the sound is happening at once, so ultimately you treat the three microphones as one. Each mic should be picking up a good sound, of course. But what matters most is how all the mics sound together."

For her debut album *Hurricane Child*, singer/songwriter Lori B. played and sang her parts simultaneously in the studio. Based on her performances, it would seem that the muse was definitely present.

ROOM TO BREATHE

Typically, acoustic guitars sound best when they're recorded in semi-"live" environments (such as rooms with wood floors), while vocals sound best—or work best in a mix—when recorded in "dry" environments. Obviously, this difference can present a problem when you want to record both instruments at the same time. The solution is to find a workable middle ground.

In my tiny studio, I put down a wood floor and position a few short wooden panels nearby to create some early reflections for the guitar. To keep the vocal track as clean as possible, though, I put up acoustic foam behind the singer.

Jones, who recorded Lori B. in a fairly large (18 × 30–foot) room with wood floors, needed to cut down on reflections. So he positioned some low-standing baffles behind the musician, to make the sound less huge and roomy. This setup worked well—at least until Lori decided she needed to stand up for a particular song. Not wanting to interrupt the flow, Jones quickly repositioned the three mics. But when she stood up, Lori B. towered above the low baffles, which brought in more room sound and subsequently made the vocals darker.

At that point Jones could have opted to set up more baffles, move to a different part of the room, or even audition other microphones to compensate for the darker vocal sound. Instead, he put Lori B.'s needs first, and, in a pinch, ended up using EQ to brighten the vocal part going to tape. "I know that's not orthodox," Jones says, "but you do what you have to do to get the sound right. The main thing is to accommodate the artist. There were several times when we had to change the setup, and there wasn't time for me to get the sound perfectly dialed in. But Lori was ready to go—and that, to me, is more important than *my* being ready. The sound can be good or bad, but if the performance is happening, no one is going to care if the microphones are less than perfectly positioned."

UNDIVIDED ARTISTRY

I still sometimes encourage a singer/songwriter to perform guitar and vocal parts separately—for example, when the song requires a click track and getting the artist to sing, play, emote, and keep steady with the click all at once is just too much to ask. But even then, it's a matter of the engineer accommodating the artist rather than the other way around.

To that end, knowing how to record acoustic guitar and vocals simultaneously is a critical, must-have skill. By accommodating the artist, you also accommodate his or her muse. This, in turn, greatly increases the likelihood of reaping one of the recordist's highest rewards: the capture of musical magic on tape.

Associate Editor **Brian Knave** *looks forward to accommodating his own muse.*

Hackers Hit the Stage

The Hub takes MIDI to the limit.

By Gino Robair with Tim Perkis

Six musicians make up the San Francisco Bay Area "computer network band" called the Hub. John Bischoff, Chris Brown, Scot Gresham-Lancaster, Tim Perkis, Phil Stone, and Mark Trayle are each armed not with a guitar or keyboard, but with a computer and synthesizer, and everything is connected to a central "mailbox" computer known as "the hub."

"The point of the Hub as a band is to create music that arises from the unpredictable behavior of interconnected computers," says Tim Perkis, who was a member of the first microcomputer network band, the League of Automatic Music Composers. Since the early '80s, the Hub has been using MIDI in a way that was never intended: as a medium of network communication between electronic musicians.

"MIDI has become the standard language that synthesizers use to speak to one another," Perkis notes. "However, the designers of MIDI gave it some built-in limitations because they had a certain way of thinking about music. For example, most of the messages sent on a MIDI cable are based on someone playing a keyboard or other controller: Note On, Note Off, Velocity, Pitch Bend, and so on. And electrically, MIDI is designed to have a single source of information, again based on the idea of one player controlling a bank of equipment. But MIDI messages can be bent and misused to implement alternative notions of musical control."

PAUL PANHUYSEN

The Hub performed at the Apollohuis in Eindhoven, the Netherlands, in 1994. The players (clockwise from left) are John Bischoff, Chris Brown, Tim Perkis, Phil Stone, Scot Gresham-Lancaster, and Mark Trayle.

DIY

The band initially was interconnected through a simple microcomputer board with serial interfaces (the original hub), which they designed and built themselves. This hub served as a common memory bank, acting as an electronic mailbox. Each player's computer could read and write messages in this common area, and the Hub's early pieces were designed to take advantage of this capability.

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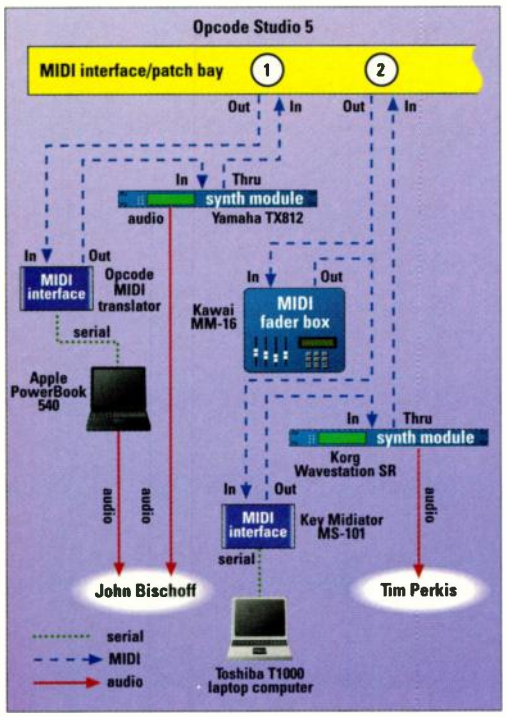


FIG. 1: In a typical Hub performance setup, each musician uses a single MIDI chain to connect with the Studio 5. Notice that Bischoff uses both internal and external sound sources.

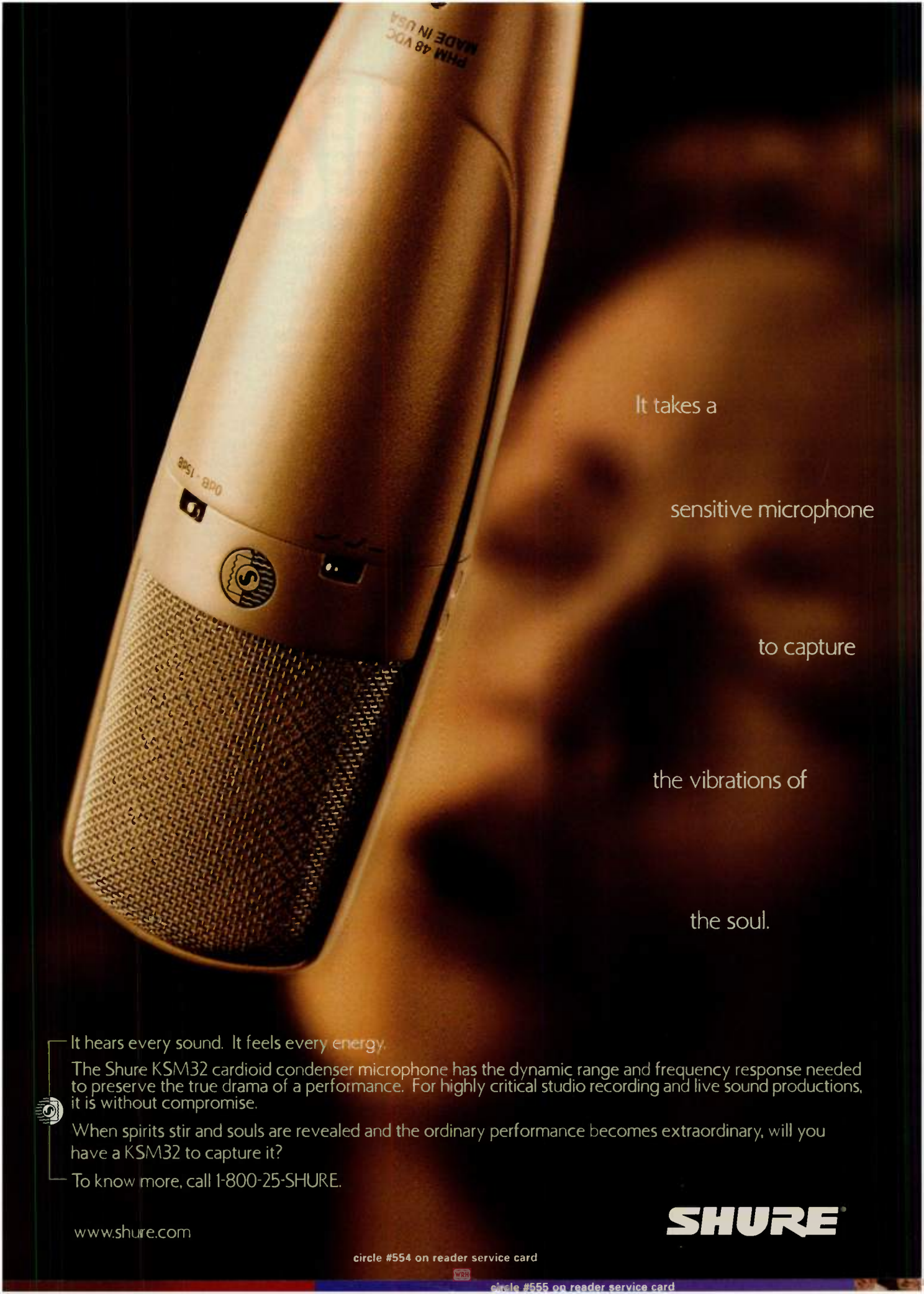
Commercial MIDI equipment has, however, matured since then. In 1990, the group began using a beta version of Opcode's Studio 5 multiport 15 × 15 MIDI interface as its hub. The resulting Hub pieces reflected the note-oriented nature of mass MIDI communication. "We used to use RS232 serial connections," Perkis says, "which offered less rhythmic synchronization than is now possible with MIDI."

It takes some doing to fool MIDI into acting as a conduit for Internet-like exchanges in which many message sources connect to many message receivers over the same network. Opcode's Studio 5 interface is perfect for the job. With a little fancy programming (provided by Scot Gresham-Lancaster), it offers enough flexibility to serve as the Hub's message router. A table of message routing and rechannelization can be set up for each of the available MIDI inputs on the device, allowing messages to be filtered, routed, and modified by message type and MIDI channel.

ROUTING

Each player is assigned a MIDI port on the Studio 5 interface (2 through 7) and a MIDI channel of the same number (2 through 7). Only six MIDI channels need to be set aside for the players because most of the musicians have single-port MIDI interfaces. The remaining MIDI channels, channel 1 and channels 8 through 15, are allocated for each player to control his own synthesizer. The MIDI Thru output on each synthesizer is connected to the MIDI Input on the Studio 5, and the corresponding MIDI Out on the Studio 5 connects to the musicians' MIDI In (see **Fig. 1**).

This arrangement allows each musician to use a single MIDI chain. The Studio 5 recognizes the channel number of an incoming message, like a mailing address, and delivers the message through the corresponding Out port to the player on the receiving end of that channel. In addition, the Studio 5 is programmed to change the channel



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REVIEWS

CLAVIA

NORD MODULAR

Virtual, analog-style modular synthesis.

By Craig Negoescu

In the distant past, analog modular synthesizers roamed the earth looking for ever larger spaces to fill with their cases of oscillators, filters, envelope generators, and miles of patch cords. These behemoths were the lumbering giants of electronic music's Cretaceous era, doomed to near extinction in a future dominated by smaller and cheaper digital technology.

Unlike the dinosaurs, however, modular synthesizers were saved from oblivion, thanks largely to companies like Clavia and retro-oriented musicians who have kept the analog faith. Clavia's Nord Modular is a hardware/software hybrid product that combines the most powerful and expressive form of classic modular synthesis with the convenience and lower cost of a digital instrument.

FIG. 1: The Clavia Nord Modular is available in three versions (top to bottom): Rack, Micro, and Keyboard. All three work with editor software to create sounds by combining virtual synthesizer modules into "patches."

THREE FLAVORS

The Nord Modular hardware is available in three configurations: Rack, Keyboard, and Micro (see **Fig. 1**). All three come in the signature Nord fire-engine red and are solidly constructed of rolled steel with functional, geometric lines.

The Nord Modular Rack is a tall, wedge-shaped unit that can sit on a desk or be mounted in four rackspaces, using the included rack ears. At the left

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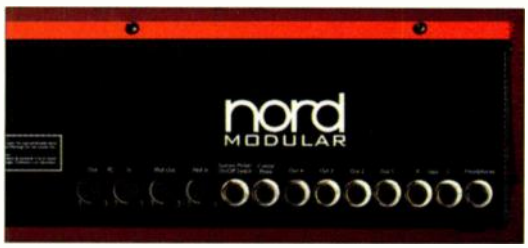

FIG. 2: The back panel of each of the three versions includes two sets of MIDI In/Out ports: one for a computer and one for other MIDI devices. It also includes one pair of audio inputs, one pair of audio outputs, and a headphone output. The Rack and Keyboard versions provide a second pair of audio outputs and two pedal inputs.

side of the front panel, there is a dedicated Volume knob, Find and Panel Split buttons, Octave Shift controls, and 18 numerically labeled knobs with corresponding status LEDs. The right side of the front panel features a backlit, 2-line × 16-character LCD, four Menu buttons, and four Patch Group buttons. A large alpha wheel is provided for data entry, along with Navigator, Shift, and Assign buttons for menu navigation. As with the keyboard version, AC power is delivered to the Rack through a permanently attached power cable, and a rocker-style power switch is located on the back of the unit.

The Nord Modular Keyboard has a 25-note, synth-action keyboard and provides the same control and display layout as the Rack in a flat, rectangular package. To compensate for the limited keyboard range, Octave Shift buttons with indicator LEDs are included at the left side of the keyboard. As a fan of Clavia's Nord Lead and its funky granite mod wheel and wooden pitch bender, I was surprised to see neither on the Nord Modular Keyboard. Lacking a pitch or mod wheel and possessing a small keyboard that is not pressure-sensitive, the Nord Modular Keyboard is not an ideal live-performance synth.

The Nord Modular Micro is a scaled-down version of the larger Rack model. The Micro lives up to its name with a staggeringly small case: it's only 8.2 × 1.4 × 4.5 inches! Despite being only slightly larger than the average paperback book, the Micro is sturdy and surprisingly heavy. The front panel is Spartan, with only four knobs, three buttons, and a 2-digit LED display. Due to its small size, the Micro cannot contain an onboard power supply, so it gets its power from an external 9-volt AC adapter that plugs into the rear of the unit. The Micro has no power switch, so it's always on when plugged in.

THE HEART OF THE MATTER

At the heart of each version of the Nord Modular hardware is a number of general-purpose DSP chips, which Clavia calls Sound Engines. These are programmed to produce various sounds by combining virtual modules (such as oscillators, filters, and envelope generators) into patches, much like the modular analog synths of yore. In this case, however, the modules are assembled and connected within software running on a PC. (A Macintosh version should be available by September.)

Different modules and patches require different amounts of a Sound Engine's processing power, which affects polyphony. In addition, each patch is limited to a single chip. The amount of processing power required by a patch is called the "patch load," which is indicated in the software's display screen. The Rack and Keyboard versions come with four Sound Engines, and four more can be added with the Nord Modular Voice Expansion Board ($799). The Micro has only one Sound Engine, and it cannot be expanded.

All members of the Nord Modular family include two sets of MIDI ports (see **Fig. 2**). One set is used to send conventional MIDI messages, such as Note On and Control Change, to and from other MIDI devices. The other set of MIDI ports is used by the controlling computer to maintain a dedicated connection to the Modular hardware for editing.

All three Modulars have left and right stereo inputs and outputs on ¼-inch jacks. The Modular Keyboard and Rack include an additional set of output jacks, as well as two pedal-input jacks for sustain and continuous pedals. All three provide a headphone jack, which is inconveniently located on the rear of the unit. As a result, it is very difficult to access the headphone jack on a Rack version if it's installed in a rack.

Of the 20 knobs on the front panel of the Modular Keyboard or Rack, only two perform dedicated functions: Master Volume and Data Entry. The other knobs are labeled only with numbers, because they can perform various functions depending on the patch assignments. Obviously, this might create some confusion, so Clavia placed a Find button on the lower left of the panel. When the Find button is pressed, any of the active knobs can be turned without affecting the settings of the patch. The display indicates the parameter associated with the knob and tells you whether the knob's position is above, below, or exactly on the currently programmed value. This feature helps prevent an unpleasant "value jump" when you turn a knob.

Above the Find button, the Panel Split button lets you divide the knobs into four groups, one for each Sound Engine. Clavia also provides a little

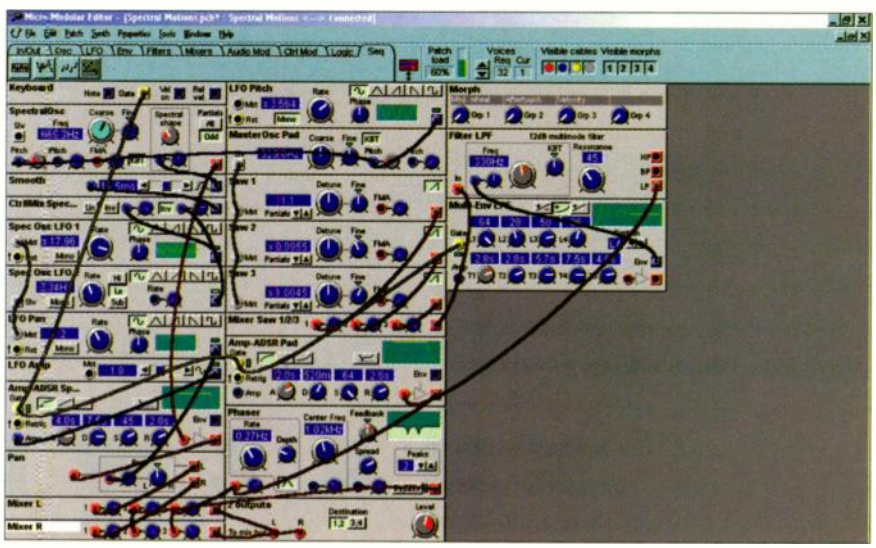

FIG. 3: This patch uses 60 percent of a Sound Engine's patch load. Notice the patch cords of different colors, which indicate their function.

For manufacturer information, please see Contact Sheet, p. 163.

extra room around the knobs so you can label them. If you don't want to mark up your pride and joy, you can use the editor software to print out a formatted patch blank that shows the parameter assigned to each knob.

FACTORY SOUNDS AND MORE

Out of the box, the Nord Modular is ready to play, with 100 factory presets. These presets are a versatile lot that exploit many features of the machine. The first factory patch, Welcome, takes a shot at the THX logo, using the Modular's considerable oscillator power. The next group of preset patches dives right into the extensive step sequencing and layering abilities of the Modular. Hearing these, you would swear you were using a drum machine and few synths, but it's all done with a single patch.

The D-fever patch knocks off Georgio Moroder with disturbing accuracy. Swing Groove layers funky analog drums and bass with various groove controls on the knobs. Phased Drummer drops a beat while you play bass and fiddle with the phaser controls.

In other memory locations, you'll find FM and other synth basses and some solo sounds showcasing the characteristic "growl" of Clavia's popular Nord Lead. The rest of the factory sounds are a mixed bag of analog, FM, and additive bells, drum kits, brass, keyboard pads, and organs. Overall, it's a well-balanced set and a great place to start exploring the sonic possibilities of the Modular.

But even a Nord neophyte need not be limited to the factory presets. Over 2,000 patches are available at the Clavia Web site, not to mention a growing list of enthusiast sites. Homebrew patches are often uploaded and traded online. Logging onto Clavia's site connects you to a network of hardcore Nord Modular hackers and their wares.

In fact, some of the most impressive patches I have encountered are free downloads from private owners. Many of their efforts have been incorporated into the factory banks, reinforcing the sense of a Nord Modular online community. Also among the many patches available at the Clavia site are 900 converted Nord Lead 2 patches and 300 DX-type FM patches modified to run on the Modular hardware.

PROGRAMMER'S HEAVEN

Playing around with the factory presets and other downloaded sounds is a lot of fun, but it only scratches the surface. Creating and modifying patches in the software editor is where the full potential of the system comes to life.

The Nord Modular editor software ships on two floppy disks and is designed to run under Windows 95, 98, or NT on a 90 MHz Pentium machine or better. For this review, I ran the editor software on a 400 MHz Micron Millennia under Windows 98 using a simple MIDI interface. The editor ran smoothly and rarely crashed.

Installing the editor software is a snap; an automated installer does most of the work for you. Clavia assumes that you already have a MIDI interface on your computer, so some users might need to install MIDI hardware and software before proceeding.

When the editor software is launched with the Modular connected, a dialog box informs you that the Modular

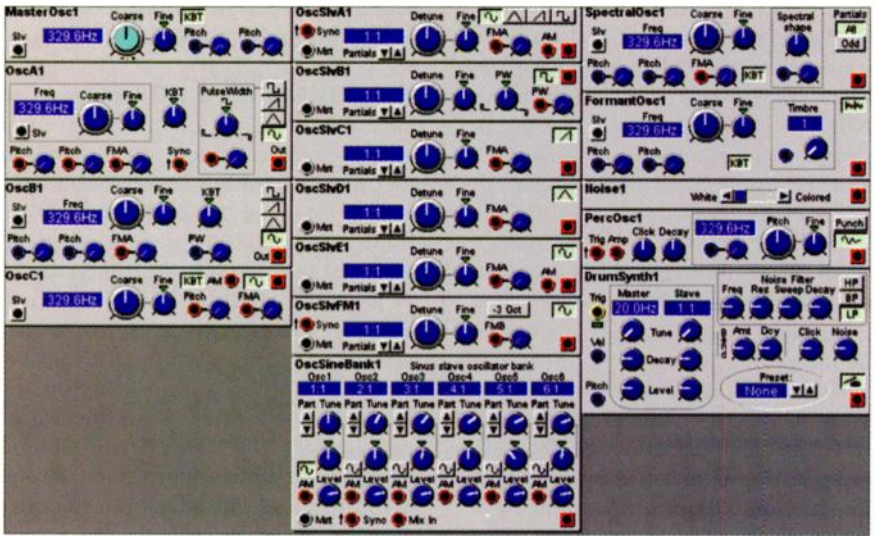

FIG. 4: The Oscillator group provides 14 different oscillator modules, many of which can be slaved to a single master oscillator to conserve patch load and thus maximize polyphony.

hardware has been found. At this point, all patches in the synth are automatically uploaded into the computer and displayed in a "floating" onscreen window. Double-clicking on any patch in this window opens it in the Patch window (where the patch is actually edited) and activates it in the synth.

Right-clicking on a module in the Patch window brings up a menu with Rename, Help, and Delete options. Renaming a module is a nice way to document your programming and helps keep track of identical modules with different purposes in a patch. Left-clicking on a patch knob or switch opens a small window showing the value or state of the item. Right-clicking on a patch knob brings up a menu that lets you zero the knob, restore it to its prior value, or assign it to a physical knob, Morph Group (explained shortly), or MIDI Control Change message. This menu also provides a comprehensive onscreen help system, complete with links for terminology, so there is no need to fumble through the thick manual.

For beginners, the editor software contains templates for simple module combinations that drop into the Patch window "preassembled" for experimentation and use. For more advanced users, the process of building a patch from scratch starts with selecting a new patch from the File menu. This opens a blank Patch window, the *tabula rasa* for your synthesizer experimentation.

A wide variety of modules is available, separated by tabbed categories in the Module menu at the top of the screen. You can grab the modules from the menu and drag them down into the Patch window. Once you have modules in the Patch window, you link them with virtual patch cables (see **Fig. 3**). Just like their hardware equivalents, these patch cables connect output to input and modulator to destination. The editor software provides four types of patch cables, distinguished by color and function: red cables for audio signals; blue cables for control signals from envelope generators, LFOs, and so on; yellow cables for logic signals used in triggers and clocks; and gray cables for master-slave connections (discussed later) and miscellaneous signals.

Making a connection between modules is as simple as clicking within an onscreen output jack and dragging the mouse to the destination jack. The cable type is determined by the nature of the originating connection. Right-clicking on a connected input or output opens a menu that lets you reassign cable types. Removing or reassigning either end of a cable is accomplished by double-clicking on the connection and dragging it to a clear area of the Patch window to break the connection, or to a new jack. The editor software does not prohibit the creative misuse of signal routing; just about any output can be patched to any input, with varying results.

After several cables have been connected, you begin to see the classic "cable spaghetti." This not-so-tasty concoction can obscure knobs and displays and generally get in the way of things. Thankfully, the software addresses this issue in a variety of ways. Onscreen cables can be "shaken" with a menu command or by hitting the Control key and spacebar together. You can hide all cables of a particular type with a control in the editor's toolbar. This lets you isolate the flow of audio, control, or other signals by hiding other cable types.

In addition, you can hide every cable for a clear view of all patch controls and displays. This feature is particularly useful for slower PCs running complex patches, when so much is present on the screen that the mouse cursor disappears because of the severe burden on the computer's display system. When cables are hidden, their connection points are filled in with colored dots corresponding to the type of cable being used.

MONDO MODULES

With over 100 modules, and more being developed, the Nord Modular can

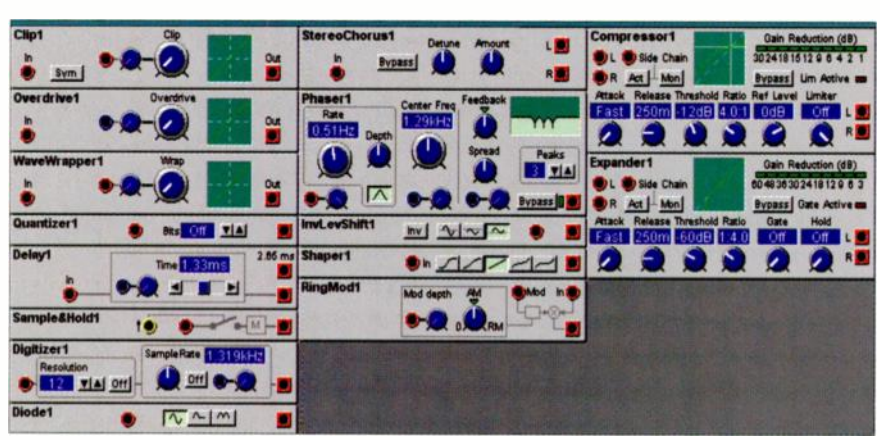

FIG. 5: The Audio Modifier group includes a variety of effects and signal processors, but there are some obvious omissions, such as reverb.

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Nord Modular Editor Software Minimum System Requirements

90 MHz Pentium or equivalent; 5 MB hard-disk space; Windows 95/98/NT; 800 x 600 screen resolution

claim the prize for maximum flexibility among the virtual analog synthesizers currently available. All modules fall within ten groups: Input/Output, Oscillators, LFOs, Filters, Envelopes, Audio Modifiers, Sequencers, Mixers, Control Modifiers, and Logic.

The Input/Output group includes ten modules for making real-world connections to the Modular hardware. The input modules provide a link from external MIDI devices, and output modules connect the patch outputs to the Nord's physical audio outputs. Every patch needs some sort of input and output, so it's nice to see that most of the modules from this group require little or no patch load.

This group also includes the Morph Group module, which lets you link up to 25 parameters and control them with one knob. A well-programmed set of Morph Groups can create an amazing range of sounds using just a few knobs.

The Nord Modular editor software ships with 14 types of oscillators, from vintage-style analog modules to more exotic digital hybrids (see **Fig. 4**). The analog-oscillator modules typically have a full complement of waveforms: sine, triangle, sawtooth, and variable pulse. They provide coarse and fine pitch adjustments, variable keyboard tracking, and a battery of control inputs.

Oscillators come in two basic types: master and slave. Master oscillators are more comprehensive, but they require more patch load. On the other hand, slave oscillators are simpler, but they require much less patch load. Any number of slave oscillators can be controlled from one master oscillator, which provides a bigger sound while preserving polyphony.

For more unusual waveforms, the Spectral Oscillator produces odd or even overtones that can be shaped into a variety of hard, biting sounds. The Formant Oscillator is capable of 127 different resonant timbres that are well suited to producing various vocal and resonant "body" textures.

The Nord Modular is ready to play, with 100 factory presets.

Combo oscillator modules combine features to save programming time and patch load. The Percussive Oscillator is a compact, efficient module that incorporates a decay envelope, mixable click, and punch control for a multitude of uses from synth drums to punchy bass sounds.

For drum machine programming, the Drum Synth module is a real time saver. It includes 30 factory presets covering a broad range of sounds. Familiar-sounding presets include Roland TR-808–like kicks and snares, some Simmons-like toms, and a couple of my favorites from the old Roland CR-78. Of course, you can also tweak the module's 14 knobs to alter pitch, envelope, click, filter, and noise to create your own percussion sounds.

Additive programming gets a boost with the Oscillator Sine Bank module. This module is a six-pack of sine waves, summed to a single output with individual tuning and level controls. The Oscillator Slave FM module works well for re-creating classic Yamaha DX-type sounds.

The Nord Modular comes with a full complement of LFO modules that feature selectable frequency ranges and waveform outputs, phase control, keyboard tracking, and visual displays of the output waveform. The LFO category also includes a bpm-based Clock Generator module and several stepped-pattern generators for random time-based effects.

For comprehensive tonal shaping, the Nord Modular has 11 filter modules. The basic filter modules come in static lowpass and highpass varieties. There are also several multimode filters with independent highpass/bandpass/lowpass outputs, variable cutoff and resonance, and slopes from 12 dB/octave up to 24 dB/octave. The Classic LP Filter module is a lowpass filter that is designed to evoke a classic

Nord Modular Specifications

	KEYBOARD AND RACK	MICRO
Audio Inputs	(2) ¼" unbalanced	(2) ¼" unbalanced
Audio Outputs	(4) ¼" unbalanced main; (1) ¼" stereo headphone	(2) ¼" unbalanced main; (1) ¼" stereo headphone
Additional Ports	(2) ¼" pedal inputs (sustain, continuous); (2) MIDI In, (2) MIDI Out	(2) MIDI In, (2) MIDI Out
Polyphony	depends on patch complexity and number of installed Sound Engines	depends on patch complexity
Multitimbral Parts	4	1
Sound Engine	DSP-based, modular subtractive synthesis	DSP-based modular synthesis
Hardware Controllers	Keyboard: 25-note, unweighted keyboard (no Velocity or Aftertouch); Rack: None	none
Programs (ROM/RAM)	0/100	0/100
Dimensions	Keyboard: 18.6" (W) x 10.4" (D) x 3.5" (H) Rack: 16.6" (W) x 6.9" (D) x 4.6" (H)	8.2" (W) x 4.5" (D) x 1.4" (H)
Weight	Keyboard: 14.5 lbs.; Rack: 10.5 lbs.	4.5 lbs.

The biggest thing in samplers wouldn't even fit on this page

- Up to 128 Voice polyphony
- Up to 256 Mb RAM
- Up to 16 outputs
- Multi-channel Digital I/O opt.
- .WAV File Format

The complete list of features is too massive to print here . . . so check out the big picture at your local Akai dealer.

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synthesizer-filter sound. It's a little smoother, with less of a "growl" to it, than previous Nord filters.

Some filter jobs are just too complex to build from scratch. For example, it's possible to build a vocoder from discrete modules on the Nord, but only the most dedicated programmers would take the time to do it, and the result would likely exceed the available patch load. Luckily, Clavia has provided a 16-band Vocoder module that requires no effort to use. Using nearly half the total available patch load, it's a real processor hog, but at least there is still enough power left to build various module groups to plug into it. And, by assigning the two mono input signals from the Nord Modular hardware, the module operates as a stand-alone hardware vocoder.

The Vocal Filter module is a complex filter designed to emulate a vocal tract. The vowel sounds *a, e, i, o, u, y, aa, ae,* and *oe* can be assigned to three different positions in the filter, and the Vowel Navigator knob sweeps through them. This module is responsible for some of the most amazing patches I've heard. A particular favorite of mine is Baby I Love You, in which, through some very clever programming, the Nord speaks a complete sentence.

The editor software provides six types of envelope generators, ranging from a simple attack/decay to a sophisticated multistage module. All the envelope-generator modules benefit from a handy display that shows the envelope's current shape. Several of these modules also have modulation inputs to the envelope stages.

The software provides the ability to do some sweetening with the Audio Modifier modules (see **Fig. 5**). The Clip, Overdrive, Signal Shaper, and Wavewrapper modules are great for shaping and distortion effects. The Stereo Chorus and Phaser modules have good sound quality, and the Phaser provides comprehensive real-time control.

Kudos to Clavia for including two modules that provide real-time resampling. Patching an external signal into the Digitizer or Quantizer module is good for hours of deconstructivist fun. Ring modulation, diode processing, amplitude modulation, and compression/expansion modules round out the Audio Modifier Group. Notable omissions from this group are reverbs and more delay-based effects.

Back in the early days of synthesis, a modular synth might include a primitive sequencer module that would accept a clock input and step through a series of control voltages set by knobs. Producing useful musical results took a high degree of patience and a steady hand at the controls.

The Nord Modular also includes this sort of step-sequencing capability, but luckily, it's not exactly like the old days. Four different sequencer modules are available, each with its own specialty. All of these sequencers feature 1 to 128 steps of sequencing, but a maximum of only 16 steps have visible LED indicators and controls. Switchable looping is available on every type of sequencer as well as a dedicated Clear button for erasing programmed steps. The sequencers are frugal when it comes to patch load; none requires more than two percent.

In the Mixer Group, Clavia provides modules to combine the audio and control signals in a typical patch.

CLAVIA
Nord Modular synthesizer
Keyboard $2,795
Rack $2,495
Micro $899

FEATURES ■■■■■
EASE OF USE ■■
QUALITY OF SOUNDS ■■■■
VALUE ■■■■
1 2 3 4 5

PROS: Nearly unlimited possibilities for generating analog sounds. Over 110 modules and growing. Comprehensive Windows editing software runs on most PC hardware. Plenty of time-saving and help features in software. Ability to process external sound. Huge public-domain library of patches.

CONS: Limited polyphony. Lack of pitch and mod wheel on Keyboard model, and keyboard is not Velocity or Pressure sensitive. Rear-mounted headphone jack can be difficult to access. Modular environment might be intimidating to users without prior experience. Macintosh editor software not available yet.

CIRCLE #437 ON READER SERVICE CARD

Three- and eight-input mixer modules are available for summing audio and control signals. Each includes a row of input jacks with corresponding level knobs, and both are basic, single-output mixers without a master-gain control. A stereo-output mixer with individual channel pans would be a nice addition here, because many effects and patches are designed to take advantage of the stereo outputs on the hardware. Beyond basic mixing functions, there is a host of other useful modules: panning, faders, crossfaders, signal switches, and other level-control modules. Most of the mixer modules accept control inputs for dynamic effects.

The Control Modifier and Logic groups include unglamorous but essential ingredients for making more complex patches. Rarely requiring more than one percent of the patch load, these tiny modules provide the "glue" for various control and modulation setups. Some of the available modules are comparators, dividers, scalers, quantizers, portamento, and logic processors.

▼

In a studio setting, the Nord Modular Rack is ideal.

THE RIGHT MODULAR FOR YOU

By now, you might be asking, "which Nord Modular is right for me?" The Keyboard version has it all: 4-part multitimbral operation, upgradeable polyphony, lots of assignable knobs, and a keyboard, which helps when performing or entering notes and chords into the sequencer modules. If you want to operate everything in one small package, the Nord Keyboard is the way to go.

In a studio setting, the Nord Modular Rack is ideal. The elimination of the keyboard also brings the price down a bit, making the Rack a more economical choice without sacrificing power. Like the Keyboard, the Rack can be upgraded for more polyphony.

The Modular Micro is also a compelling choice. It's very compact, nearly bulletproof, relatively inexpensive, and compatible with patches produced on the larger models. On the downside, the Micro is not multitimbral, and it has less polyphony. For synthesizer experimentation, sound effects, and generating sophisticated patches for sampling on a budget, the Micro will meet your needs. It's also quite useful as an esoteric, stand-alone effects box.

CONCLUSION

The Nord Modular covers the spectrum of analog and digital sounds well, with a large variety of modules. Users who have honed their programming skills on vintage modular equipment will feel right at home with the layout and function of many of these modules. However, those who are new to modular synthesis might have a difficult time with Clavia's dense interface, which offers awesome flexibility at the inevitable cost of occasional confusion.

No matter which flavor you choose, the Nord Modular is probably one of the most "hands-on" synthesizers you will own. It demands a certain level of skill to create or modify patches. If you do not have some history making analog-synthesizer patches, don't expect to jump into a blank patch and make magic. Like a traditional modular synthesizer, programming can be a tedious but rewarding experience.

The Nord Modular is at its best when used as an analog hacker's toolbox. Of course, it also can be used as a mainstream lead or pad machine in a MIDI rig, but you need to stick to simpler patches if you want to maximize polyphony. The Modular really shines when producing sounds beyond the scope of conventional analog synthesizers.

The greatest benefit of the Nord Modular is its integration with a Windows PC. For most project studios and music professionals, a computer is an essential component for composing and sound design. By using powerful DSP hardware to play sounds and taking advantage of inexpensive PCs, Clavia can reproduce the power and flexibility of an analog modular synthesizer at a fraction of the cost (and size) of a real one. With its powerful and flexible editor software and an open architecture that can easily accept upgrades, the Nord Modular is an evolving instrument that will produce some amazing sounds for years to come.

Craig Negoescu *(www.frogdesign.com) requests that everyone stop making cool synth stuff for a while so he can get some sleep.*

GT ELECTRONICS

AM52 AND AM62

These quality studio condenser mics complete the Alesis signal path.

By J. J. Jenkins

GT Electronics, the new microphone company on the block, was formed last year when the Alesis Corporation acquired the electronics division (including microphone, amplifier, and signal-processing technologies) of Groove Tubes, makers of vacuum tubes and, formerly, microphones. GT Electronics' first offering is the AM series, which consists of four large-diaphragm studio condensers: the Class A FET (solid-state) AM51 and AM52, and the tube AM61 and AM62.

I got my first glimpse of the new GT mics at the San Francisco AES (Audio Engineering Society) show in September 1998. It was hardly an optimum venue for critical listening, but even amid the din of the exhibit floor the mics sounded good enough to pique my interest.

The two solid-state GT mics utilize standard 48V phantom power. The tube models get their juice from a stand-alone power supply that comes standard with the mics, along with a 20-foot cable equipped with a Neutrik 6-pin XLR connector. Both the AM61 and AM62 employ the Groove Tubes GT5840M vacuum tube for the preamp stage. All four models have switchable 10 dB pads and 80 Hz bass rolloffs.

For this review, I tested both a solid-state and a tube model: the AM52 and the AM62. (The AM51 and AM61—$549 and $999, respectively—are basically the same mics but without the switchable polar patterns; it's probably safe to assume that they sound virtually identical to the multipattern mics in cardioid mode.)

The AM52 provides cardioid, figure-8, and omnidirectional pickup patterns. The AM62 provides the same patterns, as well as supercardioid. (A separate switch on the back makes the change from cardioid to supercardioid.) Tighter than cardioid, the supercardioid pattern is useful when you want to record several instruments in close proximity to one another.

REAL BEAUTIES

I was immediately impressed with the look, feel, and texture of the AM52 and AM62. The solid brass bodies are elegantly contoured and beautifully finished, the AM52 in matte black and the AM62 in satin nickel. I don't know about you, but I always get a better performance out of musicians when the mic looks expensive.

Both mics come with a sturdy, lockable, foam-lined flight case and a plastic hard mount for securing the mic to a stand. The AM62 also comes with a lightweight but well-crafted shock mount. The design is simple yet clever, incorporating a rotating cam to lock the mic in place.

The manual that comes with the GT mics is exceptionally good, despite—or some might say because of—the fact that it's fairly thin. It's well written and informative (a rarity), containing tried-and-true miking techniques and explanations.

GROOVIN' ALONG

During the review period, I had several sessions going on at the same time. (I tend to overbook myself.) One of them was a rock session, for which I had tracked the drums to a click but had later decided to replace all the snare tracks. To this end, the drummer brought in several vintage snare drums. We were very particular about capturing the tonal quality of these unique instruments (which included '50s Gretsches and '70s Ludwigs, though we chose the '60s Premier "Hi-Fi" for that classic British sound).

I positioned the both AM52 and AM62 (each in cardioid mode) about eight inches away from the drum at a 45-degree angle to the batter head and went into the control room to hear how they sounded. The AM62 was distorting a bit, so I kicked in the pad. That got rid of the distortion, but I still wasn't thrilled with the tone. So I decided to back the mic off a bit and try it again with the pad disengaged. This improved the sound somewhat, but it was still a bit bright for my taste. Basically, the AM62 just wasn't the right mic for the job.

The AM52, however, sounded good right off the bat. It was a bit noisy, but the tone it captured was great—just like the drum sound I heard in the room. Next, I auditioned the AM52's various polar patterns. As it turned out, the figure-8 pattern was marvelous at capturing the reverberant acoustic characteristics of the drum room. We overdubbed the snare part with the figure-8 pattern and it sounded great.

After completing the snare track, I decided to audition the AM52 and AM62 on some other drums and percussion, since I already had the mics set up. The AM52 excelled on toms and congas. Cymbals and bell-type percussion sounded a little flat, though. The high end was there, but it was just a bit dull sounding. The AM62, on the other hand, delivered these sounds with crystal clarity. Compared to the AM52, it

The GT Electronics AM62 is one of four new large-diaphragm studio condenser mics available from Alesis. (Two of the mics are tube and two are solid state.) The AM62—our reviewer's favorite—is a tube model featuring four polar patterns, a 10 dB pad, and an 80 Hz high-pass filter.

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BITHEADZ

UNITY DS-1 1.2 (MAC/WIN)

This 24-bit "virtual" sampler rivals the hardware competition.

By David Battino

Are hardware samplers obsolete? If not, they're starting to look mighty redundant. Crack one open and you'll find some RAM, a disk drive or two, a handful of audio jacks, and onboard software that tells the machine how to respond to MIDI. Except for the software part, that's strikingly similar to what's inside a computer. Of course, hardware samplers also include custom chips to handle the audio processing, but today's PCs can be beefy enough to perform these same audio tasks on their own.

With *Unity DS-1*, BitHeadz has created a software sampler that surpasses its expensive hardware rivals in several important ways. I reviewed the program on a Macintosh, but a Windows version is also available, and according to BitHeadz, the two are nearly identical.

COME TOGETHER

Depending on your computer's audio hardware, *Unity DS-1* can deliver up to 24-bit resolution at sample rates of up to 96 kHz (48 kHz on the PC). On the Mac it supports ASIO-compliant products, Digidesign hardware, Sound Designer–format samples, and built-in Sound Manager audio. The Windows version—which also supports ASIO—requires a DirectX-compatible sound card. Both versions support SoundFont 2 samples. Because *Unity DS-1* doesn't have to squeeze samples through the low-end SCSI ports that plague most hardware samplers, its loading time is blindingly fast: I loaded an 18.7 MB bank of sounds in under four seconds. By comparison, it took a Kurzweil K2000 almost a minute and a half to inhale a 16 MB bank. That's 26 times as long.

Another benefit of having your sampler right in your computer is efficiency. You don't have to buy two sets of RAM, and you don't have to scare up a free MIDI port, AC outlet, and set of mixer inputs. What's more, you can use standard file-management techniques to organize and back up the sampler's data. And because *Unity DS-1* reads *and* writes AIFF files (on the Mac), exchanging sounds with sequencers and audio editors is a snap. (The Windows version can read but not write AIFF files, but both versions can also read and write WAV files.)

Unity DS-1
Minimum System Requirements
Mac: PowerPC 120 (G3 recommended); 32 MB RAM; Mac OS 7.6.1 (Mac OS 8 recommended); CD-ROM drive
PC: Pentium 200; 32 MB RAM; Windows 95 or 98; DirectX-compatible sound card; CD-ROM drive

One of *Unity DS-1*'s coolest features is its ability to "play" its output to disk as a stereo audio file. Import that file back into the sampler, assign it to an oscillator, and you can build up massive layers of sound. Thanks to this resampling capability, you never really run out of polyphony. However, turning any computer into a responsive, dependable musical instrument is a challenge, even for clever software like this. So, what's *Unity DS-1* really like?

WHAT MAKES IT TICK?

If you've used *Retro AS-1*, BitHeadz's "analog" software synth (reviewed in the February 1999 **EM**), you'll find *Unity DS-1* familiar. *Unity DS-1* essentially adds sampling, keymapping, and a waveform editor to the *Retro AS-1* engine, along with twice the polyphony. *Unity DS-1* provides two stereo oscillators per voice instead of *Retro AS-1*'s three mono oscillators, but you still need a powerful computer to run it. BitHeadz has made another improvement in that the signal-flow diagram used in *Retro AS-1* has been enlarged and made editable in *Unity DS-1* (see **Fig. 1**).

The *Unity DS-1* CD-ROM contains more than 250 MB of samples, organized into 79 banks of programs. (A bank file contains samples, keymaps, *and* programs, so you don't have to mess with multiple file types, as in Digidesign's *SampleCell*.) To play *Unity DS-1* as a real-time MIDI sound module, you launch the Serial Input program, which directs data from your MIDI interface to the sampler engine. If you're running OMS or FreeMIDI and have configured them to recognize *Unity DS-1*, you can launch OMS Input or FreeMIDI Input instead. Start playing your MIDI controller, and you'll hear *Unity DS-1*'s 8 MB General MIDI bank.

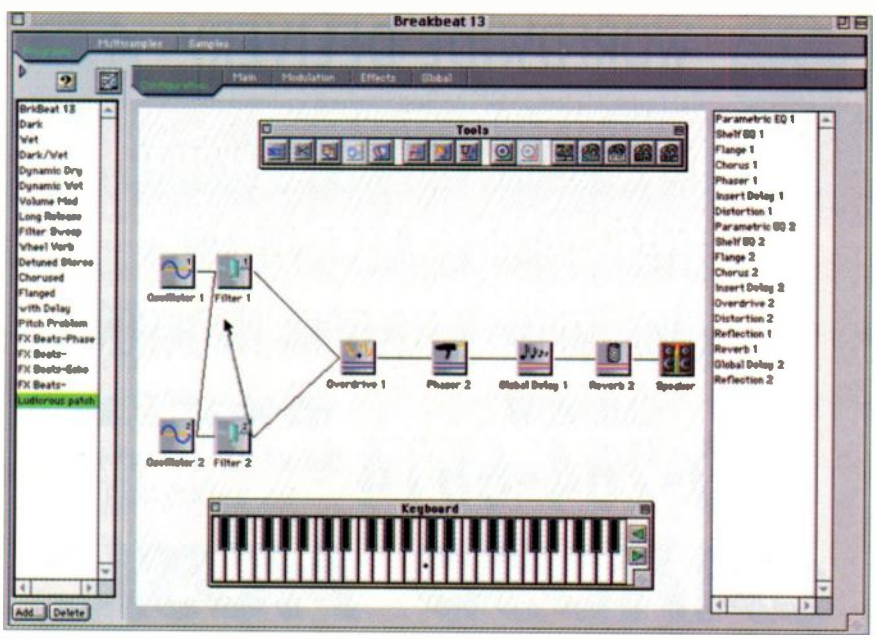

FIG. 1: *Unity DS-1* offers two stereo oscillators and two filters per voice; the filters can feed back on each other. Up to four effects can be dragged into the signal chain from the list at right.

I was astonished to find that, in *Unity*, latency—the scourge of all software synthesizers—was almost nonexistent: there was only the tiniest delay between pressing a key and hearing the sound. Even better, that delay was consistent, so I soon forgot about it. My initial skepticism about the viability of a software-based sampler disappeared. (Older operating systems may select a larger buffer size for Sound Manager, which may slightly increase latency, but because *Unity DS-1* is a RAM-based sampler, slower computer speeds do not affect latency.) Your audio hardware and its drivers, however, may increase latency, so download the free *Unity DS-1* demo from BitHeadz's Web site, and check it out on your system before you buy. *Unity DS-1* allows you to adjust several of its parameters to optimize the sampler's performance for your setup (see **Fig. 2**).

Unity DS-1 lets you access up to 16,384 programs over MIDI. You just send it a Controller 32 and a Controller 0 value to select the bank, followed by a Program Change command to load the bank and select the program. If you want to see the names of the banks and programs, though, you'll have to fire up one of the companion applications that contains bank and program menus—Keyboard, Mixer, or MIDI Processor. With a bit of preparation, you can make the names show up within an OMS or FreeMIDI-compatible sequencer, too. That makes program selection even easier. If you want to edit the sounds, you have to launch the Editor application.

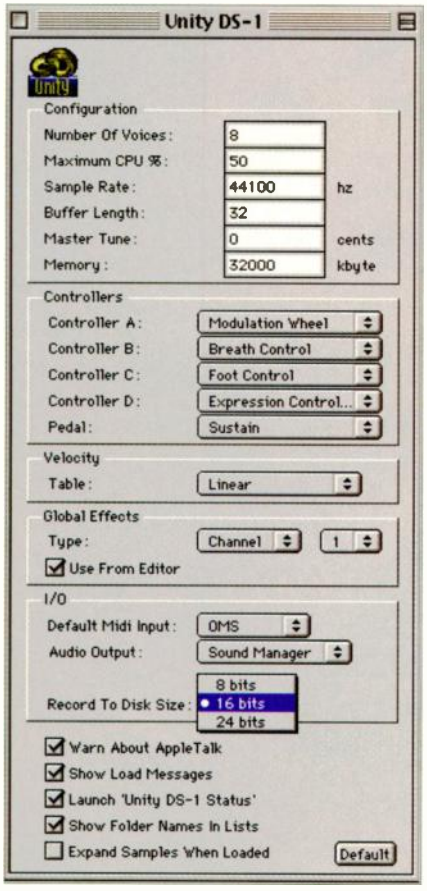

FIG. 2: Optimizing *Unity DS-1* involves tradeoffs. A shorter buffer length speeds up response but reduces polyphony. Selecting "Expand Samples When Loaded" increases polyphony but doubles RAM requirements.

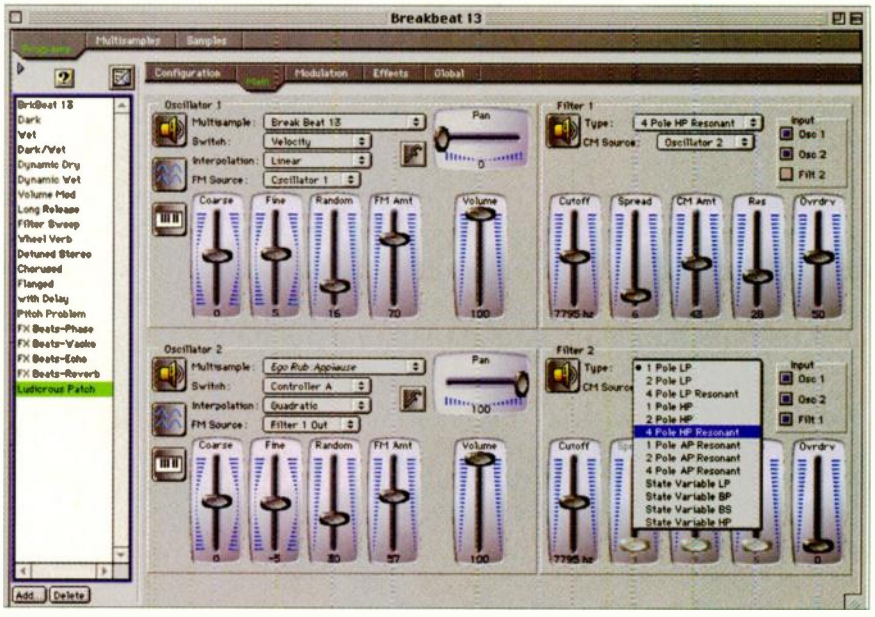

FIG. 3: *Unity DS-1* offers 13 types of filters, and they sound pretty good. All slider functions can be controlled via MIDI in real time.

As you can see, *Unity DS-1* (like *Retro AS-1*) is actually a suite of programs. In addition to the seven already mentioned, there are utilities that display CPU usage, process patch files for efficiency and PC compatibility, and prepare patch-name lists. This modular design lets you use the minimum amount of RAM for each musical task, but it's annoying nonetheless. Imagine if a hardware sampler came in ten pieces.

Then again, most hardware samplers max out at 128 MB of RAM, whereas *Unity DS-1* can address as much RAM as your computer can hold—minus the portion used by the operating system and the program itself. That portion can be significant, though: by disabling nonessential extensions and control panels, I pared my OS down to about 12 MB. The *Unity DS-1* engine requires 7 MB, and the Editor application needs an additional 4 MB along with enough RAM to hold the largest bank you plan to edit. So to edit and play a 16 MB bank, I had to allocate 55 MB (12 + 7 + 4 + 16 + 16), and I hadn't even launched my sequencer yet.

On the bright side, *Unity DS-1*'s load time is so fast that you may not need as much RAM as you would on a hardware sampler, in which you'd typically want all the samples for a session to be loaded and waiting. The manual claims that *Unity DS-1* can also load new samples into free RAM while playing old ones, but I got clicks when I tried to do that.

Speaking of the manual, it's a 291-page PDF file, so set aside another 6.5 MB of RAM to run Adobe *Acrobat*. Because *Unity DS-1* is so complex (and its online help is so weak), you'll want the manual at your fingertips for quite a while. Although lengthy, it's quite clearly written.

SOUND ASSORTMENT

Unity DS-1 comes stocked with a wonderful variety of samples, and all but the five Bill Bruford demo banks are yours to use freely. You get piano, Rhodes, harpsichord, acoustic guitar, viola, cello, trumpet, flute, sax, loops, effects, and *Retro AS-1* waveforms. Many banks include multiple programs that show off *Unity DS-1*'s extensive modulation capabilities. You'll also find a colossal helping of Sound-

For manufacturer information, please see Contact Sheet, p. 163.

Font conversions, which were originally designed for 4 MB Sound Blaster cards. They're endearingly short and gritty.

I really like the "construction kit" banks, which break out multitrack grooves onto adjacent keys. By simply activating the record-to-disk feature and wiggling a few fingers, I could produce complete stereo backing tracks ready to blow into my sequencer and embellish with MIDI overdubs. (Of course, I could also sequence the entire performance and render it to audio later.) However, I was disappointed with quite a few of the other sounds. The decay envelopes are often unnaturally short or long, and multisample split points are sometimes too obvious. But hey, it's a sampler, so instead of bellyaching about the freebies, let's load up some new sounds. To do that, we launch...

THE EDITOR

Reflecting the structure of a *Unity DS-1* bank file, the Editor application has three main pages: Samples, Multisamples, and Programs. Within the Programs page are five subpages: Configuration, Main (oscillator/filter), Modulation, Effects, and Global settings (see **Fig. 3**). Fortunately, you can quickly navigate between pages by pressing computer keys. In fact, I would like to see key commands for many more functions. In particular, it is disappointing that the Enter key can't be used to input numeric values; only the Return key works. It would also be nice if you could double-click on a slider to zero out its value. My biggest interface gripe, however, is the lack of pop-up help. With hundreds of adjustable parameters, *Unity DS-1* would be much easier to use if you could hold down a key, click on any field or slider, and read a succinct description of what it does.

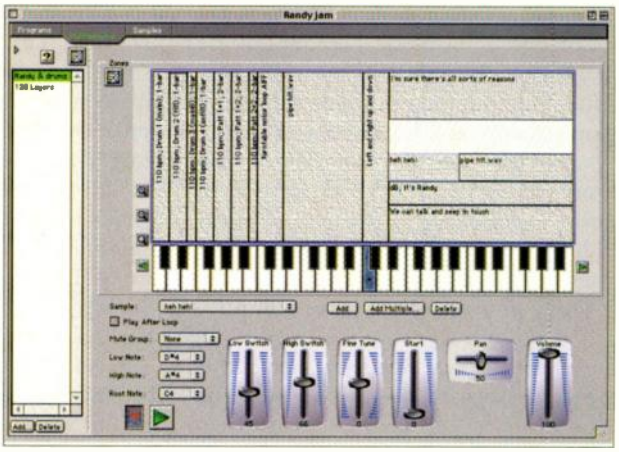

FIG. 5: In *Unity DS-1*, you can assign up to 128 samples to each key and then switch between them with Velocity.

Because the Oscillator and Modulation pages are very similar to those in *Retro AS-1*, I'll refer you to that review and focus on the Sample and Multisample pages. Suffice it to say that nearly anything can modulate anything else. There are some new sampler-friendly routings: you can modulate a sample's start point (progressively skipping attack transients as Velocity decreases, for example). You can even allow each sample in a bank to have its own set of program parameters (such as LFOs and envelopes).

Unity DS-1 excels in the number of sample formats it can read (see the table "Unity DS-1 Specifications"). You can drag and drop audio and keymap files into the Editor windows. The program can even commandeer your computer's CD-ROM drive to import Akai multisamples, loops, and programs, which is a tremendous boon. Akai program parameters aren't recognized (*Unity DS-1* just grabs the samples, keymaps, and names), but BitHeadz sells a $179 program called *Osmosis* that provides more complete file translation capabilities. *Unity DS-1* spits out the CD after each file loads, which can be irritating for those who want to import several Akai files from CD. (*Osmosis*, on the other hand, allows you to convert multiple programs or samples and does not automatically eject the disc.)

Unity DS-1's audio editor (see **Fig. 4**) offers several nice editing tools. Although it lacks modern utilities, such as time stretching, pitch shifting, scrubbing, tempo detection, bidirectional looping, and loop shifting (locking the loop length and sliding both loop points back and forth together), a workaround is provided. With one click, you can export the sample to another editor, such as BIAS *Peak*. Another handy feature, called Split Audio, chops a continuous drum groove into individual hits, à la Steinberg *ReCycle* but in stereo. I used it to separate a vocal track into phrases. You can also activate this function while recording to automatically split the incoming signal into separate samples.

Sorely missing from the audio editor is a real-time display. The program offers

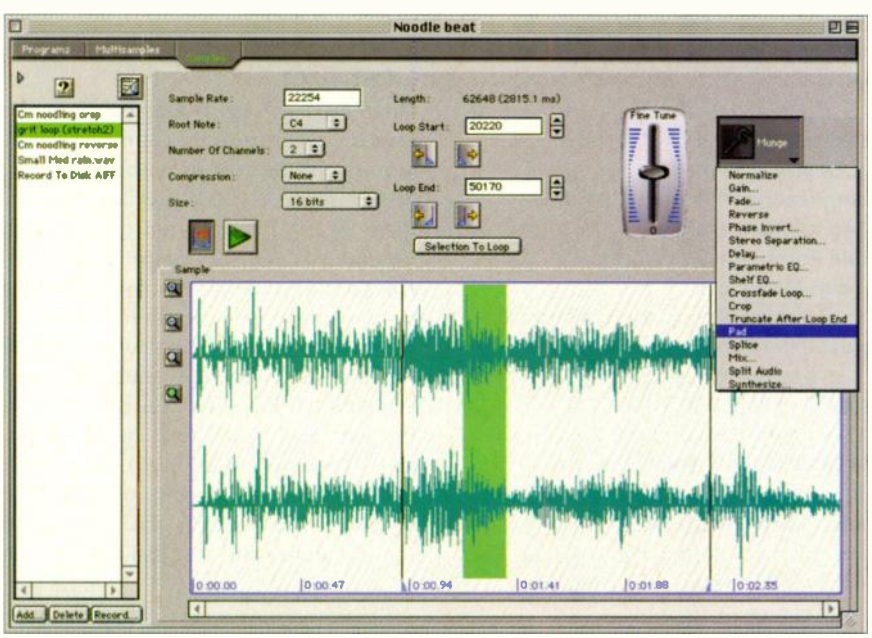

FIG. 4: *Unity DS-1*'s sample editor has nice touches, like buttons that move the loop boundaries to the nearest zero crossing. The unique Pad command keeps nonlooped samples from squealing when they reach their end.

no moving cursor to show playback position and no numerical readout of the length or endpoints of a selected region. For that matter, there's no loop-point matching window, something the Akai S1000 had a decade ago. I also would have liked a "truncate before loop" command. But on the bright side, it's rare to find a hardware sampler with an Undo button.

Once you've looped your samples, it's time to assign them to keys. The Multisample page (see **Fig. 5**) makes this process easy with one-button importation of up to 128 samples and a drag-and-tweak display. There are also four Mute Groups (exclusion groups) to force single-voice playback of zones, so, for example, you can set up a closed hi-hat sample to cut off an open hi-hat sample. A sample can also cut itself off. Velocity and positional crossfading aren't implemented, but you can program up to 128 Velocity switch points per key.

After creating your multisample, you assign it to an oscillator and start playing. (Oscillators can also play multisamples stored in other banks.) If things sound a bit dry, you may want to visit the Effects page. Unfortunately, this is one of the weakest parts of the program. The reverb—available only as a global effect, not an insert—is grainy and has a weird out-of-phase quality. The chorus is okay, but the flanger and phaser are a bit too subtle, and the distortion/overdrive effect breaks up and fizzes easily. Happily, the delays sync to MIDI clock (unlike the MIDI Processor's arpeggiator, which picks up the tempo but doesn't stay locked.)

Unity DS-1 doesn't support plug-in effects directly, but it does support *ReWire* and MAS 2.0. This allows the audio output of individual MIDI channels to be routed into separate sequencer mixer inputs, and effects can be applied there.

THE COMPETITION

Unity DS-1 shines in a desktop music environment, but a hardware sampler would be a more reliable choice if you play live. I made heavy use of *Unity DS-1*'s Reset and Update commands. This wasn't necessarily the program's fault; computers can be pretty temperamental.

A good compromise for those without an audio interface might be a sampling sound card like the E-mu APS; in fact, BitHeadz is writing the front end for the Mac version of the APS. Also, several other software samplers are elbowing in on *Unity DS-1*'s turf, including Koblo's *Stella 9000* (Mac), Seer Systems' *SurReal* (Win), Native Instruments' *Transformator* (Mac/Win), and NemeSys's *GigaSampler* (Win). The big edge that *Unity DS-1* has over all of these contenders is that it can import many more file formats—including SoundFont banks, which is why it caught E-mu's eye.

BITHEADZ
Unity DS-1 software sampler
$449

FEATURES ■■■■
EASE OF USE ■■■
DOCUMENTATION ■■■
VALUE ■■■■
1 2 3 4 5

PROS: Extremely fast sample loading. Extensive modulation capabilities. Imports numerous sample formats. Supports 24-bit audio.

CONS: Requires fast computer and lots of RAM. Complex setup. Sprawling interface. Meager online help. Rudimentary waveform editor.

CIRCLE #439 ON READER SERVICE CARD

Unity DS-1 Specifications

Polyphony	up to 64 notes (CPU dependent); 2 phase-locked stereo oscillators per voice
Multitimbral Capability	16 parts
Maximum Resolution	24 bit
Maximum Sampling Rate	96 kHz (Mac); 48 kHz (PC)
Filters	2 per oscillator, 13 types (including resonant), serial or parallel
Envelopes	unlimited number; 6-stage
LFOs	unlimited number; 6 waveforms, can sync to MIDI clock
Effects	2 inserts per channel plus 2 global effects: EQ, delay, reverb, chorus, phaser, flanger, overdrive, distortion
Sample Formats Read (Mac)	AIFF, Akai S1000/S3000, DLS, Sound Designer II, SoundFont 2, SampleCell I/II, WAV; also supports live audio recording and CD audio capture
Sample Formats Read (PC)	AIFF, Akai S1000/S3000, DLS, SoundFont 2, SampleCell, WAV; also supports live audio recording and CD audio capture
Keymaps Read	Akai S1000/S3000, DLS, SoundFont 2, SampleCell I/II
Velocity Splits	up to 128 per note

LAST BITZ

Unity DS-1 is *deep*. BitHeadz gives the user an extraordinary (and daunting) amount of control over the program. For example, you can pick among four types of pitch interpolation for each oscillator. Each successive choice provides a slightly smoother sound when a note is transposed—and requires more processing power. The program is full of these decisions and trade-offs, and after more than two months of working with it, I'm still not sure I'm getting its full potential. But for me, *Unity DS-1* has replaced a box of jacks and chips. Will it work for you? Download the free demo and see. Just try *that* with a hardware sampler.

David Battino *is the editor of* **EM***'s* Desktop Music Production Guide. *He once completed a film score using a hot-rodded Fairlight that crashed after every few notes.*

Let the "critics" tell you how easy the Spirit Digital 328 mixer is to use...

Spirit's Digital 328 represents a new way of thinking in digital console design—it bridges the gap between analog ease-of-use and digital sound quality and features.

George Petersen of* Mix *says: "There are more than a dozen entries in the 'low cost' category of digital consoles, but in terms of pricing, performance and fast, logical interface, the Digital 328 clearly sets itself apart from the pack."

Take a few moments to read what he and other "critics" say about the Digital 328. Then, go to *www.spiritbysoundcraft.com* on the web for more information. If you're in the market for an *affordable* digital console, you need look no further.

On 328's user interface:

"The 328 is a real console interface that immediately feels as close to your comfortable old analog board as you could want... the consideration that has gone into every single button, knob and interconnect is striking." – Recording

"I liked the user interface a lot, and given that the most-requested features and digital interfaces are all included, the price is excellent." – Electronic Musician

"I like this board. It has a logical interface and enough knobs for fast operation (as such it could be ideal in a live performance or broadcast situation) while its audio performance is clean enough for any recording application." – Mix

On 328's E-Strip:

"The invention of the E-Strip is a stroke of genius, [giving] instant access to all controls at once on the selected channel." – Audio Media

"The 328 is fast and intuitive, thanks in large part to its 'E-Strip' interface. There are no subroutines or hidden pages; anyone familiar with an analog console can sit down at a 328 and be working in a matter of minutes." – Mix

"With Spirit's clever E-Strip design, this digital desk has the feel of an analogue." – The Mix (UK)

On 328's equalization:

"...To my ears, this is one of the most musical sounding digital EQs I've ever heard." – Recording

"[One] of the best features of the desk: carefully tailored to provide control ranges similar to those on a top-notch analogue console, it is (dare I say) very musical." – Audio Media

On 328's effects:

"A strong selling point for this unit is the pair of built-in stereo Lexicon effects... Having quality effects in the digital domain makes for clean sounds." – Electronic Musician

On 328's automation:

"The automation is straightforward to set up and works well." – Audio Media

"Between the user setups, snapshots and dynamic automation, the 328 remembers everything except the line-input trims and 100Hz rolloff switches. It's easy to get used to this way of working." – Electronic Musician

On 328's connectability:

"Clearly, the Digital 328 provides a multitude of configuration options suitable for project studios, post-production facilities, radio stations and even live applications." – Electronic Musician

"The 328 interfaces to practically anything digital." – Recording

On 328's unbeatable value:

"All in all, the British have indeed landed with a winner. The more you use this board, the more you will discover its depth and power. With one of these consoles, you could start a musical revolution of your own." – Electronic Musician

"This mixer packs a mighty punch for $5,000 [suggested list price]. It sounds excellent, does an excellent job of untangling all the various digital formats in use, and has an excellent interface. A bold step forward in digital console design." – Recording

"I like this desk! There's nothing better out there right now than the 328." – The Mix (UK)

On 328's mic preamps:

"The mic preamps have plenty of headroom... I was surprised at the clarity of the most subtle nuances of the percussion, including the last hint of sound from the bell trees and chimes." – Electronic Musician

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LEXICON

MPX G2

The venerated reverb company rolls out a premium guitar processor.

By Barry Cleveland

When Lexicon set out to create a multi-effects processor for guitarists, it assembled a team of consultants that included legendary recording engineer Eddie Kramer; guitarists John Petrucci, Carl Verheyen, Blues Saraceno, and David Torn; and other industry hotshots. The result, after nearly two years of effort, was the new Custom Shop series of products, which includes the MPX G2 guitar effects processor and its companion pedalboard, the MPX R1 MIDI remote controller.

Readers familiar with Lexicon's MPX 1 multi-effects processor will be right at home with the MPX G2. In fact, at first glance you might even mistake one for the other. Both are single-rackspace units with nearly identical control surfaces, and their editing, routing, and programming functions are quite similar. Moreover, the G2 employs the same Multiple DSP Architecture used in the MPX 1, with a proprietary Lexichip for reverb and a second "fast-math" DSP for the other effects, ensuring uncompromised reverb quality. The G2's 24-bit A/D and D/A converters sound great (a "Soft Saturation" analog limiter can be inserted before the A/D converter) and internal processing is 32-bit, so the overall audio quality is excellent.

But the MPX G2 is not simply a guitar-oriented version of the MPX 1. The G2 has more features than I can cover in the space allotted to this review. Among other things, it offers an analog preamplifier with dedicated tone knobs (low, mid, high); a 20-second JamMan-style phrase sampler; two noise gates; a chromatic tuner; digital emulations of vintage analog effects pedals; and a unique dual-signal-path design. The digital pedal emulations can be placed in front of your amp's input or in its effects loop (using dual signal paths), in nearly any order, and reconfigured with the push of a button. If your amp has no effects loop, you can connect the G2 to your amplifier's input. And if you have no amp at all, you can use the G2 as a stand-alone preamplifier connected to a power amp or a mixer. Add the MPX R1 MIDI Remote Controller (see the sidebar of the same name), and you have a remote pedalboard that controls your digital stompboxes *and* your virtual rack effects.

GAINING CIRCULATION

The MPX G2's internal switching and routing system is very sophisticated. Centered on a module called the Dynamic Gain Block, it uses an ingenious arrangement of inputs and outputs. Effects can be placed in front of the Gain Block (pregain) or after it (postgain) on either of two discrete signal paths. Signals that have passed through the effects on the pregain path and then through the Gain Block can be routed to your amp's input via an Insert Send. Also, signals from your amp's effects sends can be fed back into the G2's Insert Returns, processed by the effects on the postgain path, and routed out the G2's main outputs to your amp's effects returns. (For a diagram of the signal flow, see **Fig. 1.**) The Insert button on the front panel triggers a hard-wire bypass, making this arrangement particularly useful.

The MPX G2 employs two gates. A mono analog gate positioned between the Gain Block and the Insert Send (see **Fig. 1a**) gates noise generated by effects in the pregain path. A stereo digital gate is positioned between the Insert Returns and the main outputs (see **Fig. 1b**). Some of the G2's gain effects are noisy, so the gates really come in handy. Both gates work quite well and outperform most others I've tried. Still, I experienced some problems with reverb tails getting cut off of sounds with subtle decay characteristics. Also, the abruptness with which the gate chops off noisy effects is problematic.

Besides acting as the Grand Central Station of the MPX G2, the Gain Block also functions as one of seven Effects Blocks (more on these later). However, the Gain Block, unlike the other Effects Blocks, is analog rather than digital, and its effects (Preamp, Split-Preamp, Tone, Crunch, Screamer, Overdrive, and Distortion) can be placed only on the pregain path.

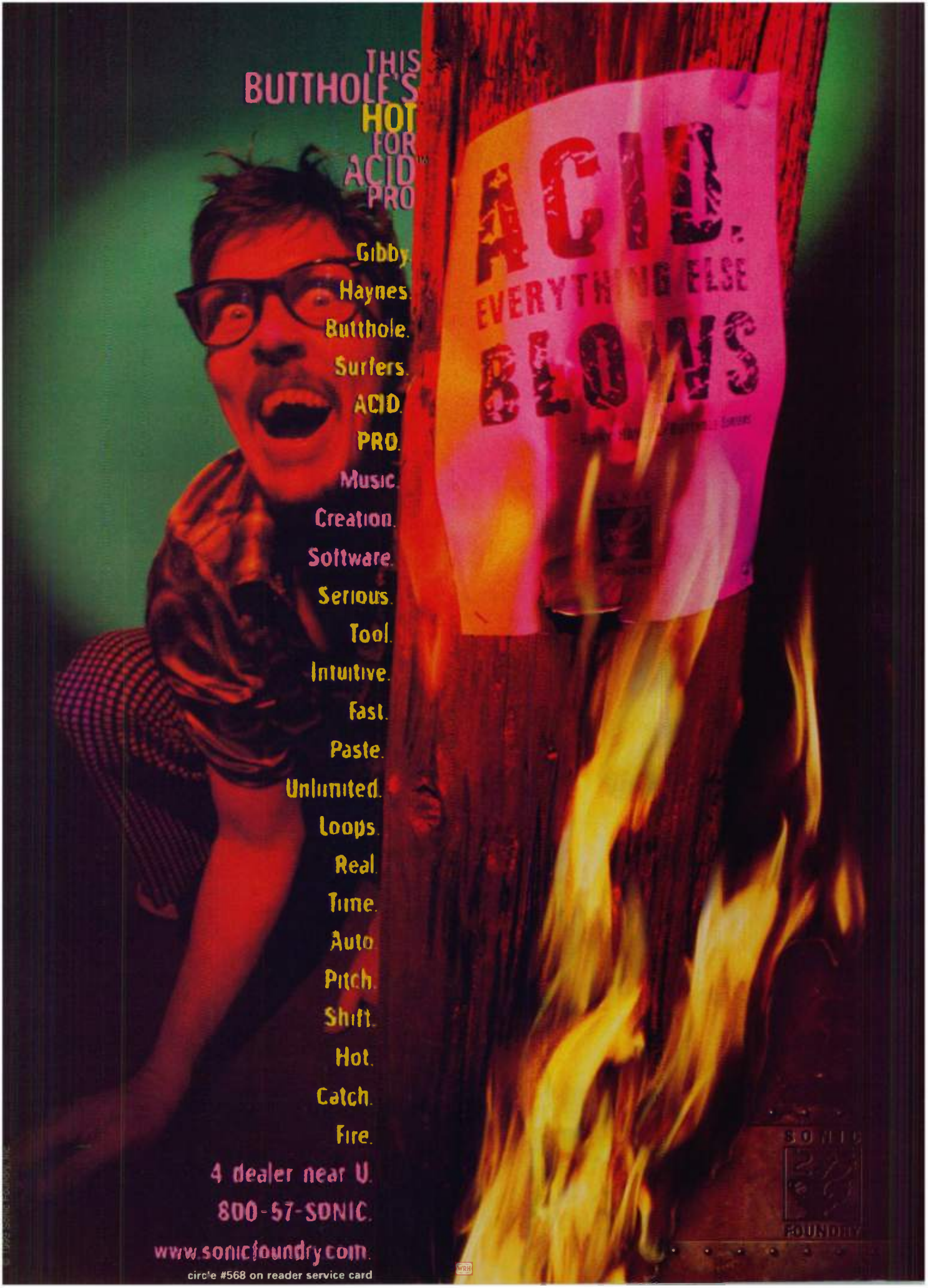
I'll deal with most of these "effects" shortly. Actually, Preamp and Split-Preamp (the latter sends a mono signal to one channel) are not effects at all. Rather, they enable the G2 to act as a stand-alone preamplifier, particularly when used with the Speaker Simulator. These offer an extraordinary amount of tone control and even a "Feel" control that supposedly mimics the power-sag envelopes of various power-amp rectifier circuits, both solid state and tube. Does that mean you can leave your amp at home? Well, maybe.

AMPLE AS AN AMP?

In the MPX G2 User Guide and in advertisements, Lexicon takes pains to make clear that the G2 is meant to complement rather than replace your favorite guitar amplifier. But the unit can also be used without an amp. Indeed, preset programs 180 through 199 are referred to as Amp Collection and include patches with such names as American Clean, AmericanMod, British '70s, and TransChorus 1. These patches are clearly intended to sound like those types of amplifiers.

Of course, it would be a tall order for any amp—especially one with a tubeless preamp—to produce all of those sounds, so some common sense is called for. If your sound is largely the

In addition to multiple effects, the Lexicon MPX G2 provides an analog preamp, a 20-second phrase sampler, two noise gates (one digital, one analog), digital emulations of vintage analog effects pedals, and a sophisticated switching and routing system.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Multiple I/O allows the MPX G2 to be used in the effects loop of a guitar amp, in the front end of an amp, or both—or even as a stand-alone preamp. The 250 ROM presets are divided to accommodate each of these three applications.

result of your playing through, say, a tweed Fender Twin, a Hi-Watt from the classic "hand-wired by young girls" period, or even a first-rate modern tube amp, the MPX G2's preamp will not be an adequate substitute. On the other hand, if your sound is less dependent on tube tone, you're looking for new sounds, or there are occasions when you'd prefer to "go direct" rather than lugging your amplifier, you'll probably find the MPX G2's preamp more than ample.

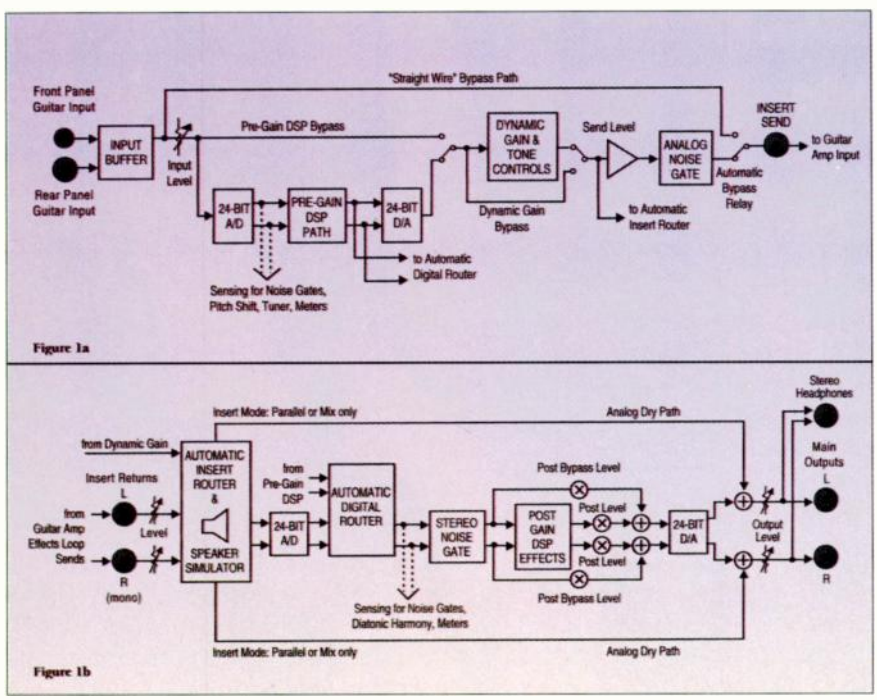
FRONT AND BACK

Considering the depth and complexity of the MPX G2, its control interface is remarkably intuitive and easy to use. On the left, next to the ¼-inch guitar input, is the analog gain section, which provides two concentric knobs for master input and output levels, and three dedicated knobs (Low, Mid, High) for adjusting tone. To the right of the display, nine small backlit buttons select the seven Effects Blocks and access the Insert and Bypass functions. Two slightly larger buttons let you tap in delay times (Tap) and activate the Glide continuous controller (A/B). To the right, a large knob and two buttons (No, Yes) are for selecting or modifying various functions and parameters. Other buttons allow you to select operational modes (Program, Edit, Store, System, Soft Row) or modify the functions of other controls (Options).

The unit's large display is divided into two windows. The first window displays program or patch numbers and tuner values, and the second has a 2-row, 16-character display for program names, effect names and values, and so forth. A green Input LED lights when signal is present at either the front or rear guitar inputs, and a red Clip LED indicates overload at those inputs or the presence of an effect in the pregain path. Similarly, an Aux In LED shows signal present at either of the two Insert Return jacks, and an Aux In Clip LED indicates overload at either of the Insert Returns or the presence of an effect in the postgain path.

There are no dedicated level meters, but you can select a very clever and useful Meter Array from the Edit menu. Its eight small, 7-segment meters simultaneously display input level, main output level, and the levels of the six DSP effects blocks. However, the array disappears when you change programs, making it more useful for troubleshooting than for general monitoring. Here's another nice touch: whenever you edit a program's rate, depth, headroom, or other dynamic value, a tiny bar meter appears in the Edit window and displays the appropriate parameter.

The MPX G2's rear panel provides a second ¼-inch input jack (overridden by the one on the front); main output jacks (both XLR and ¼-inch TRS); a headphone jack; a mono Insert Send jack; stereo Insert Return jacks (with a dedicated level pot); ¼-inch TRS jacks for footswitch and footpedal controllers; and MIDI Remote/In, Thru, and Out connectors. The 7-pin MIDI In connector serves as a Remote In when used with the MPX R1. The additional two pins route power to the MPX R1 when an external AC adapter is connected to the G2's Remote Power In jack. The G2 has an internal power supply and a detachable IEC power cord.



COURTESY LEXICON

FIG. 1: These diagrams show signal flow for the MPX G2's pregain (1a) and postgain (1b) paths.

EFFECTS

The MPX G2's 76 effects are arranged in seven Effects Blocks: Gain, Effect 1, Effect 2, Chorus, Delay, Reverb, and EQ. Only one effect per block can be active within a program, but some of the more commonly used effects are available in more than one block.

A lighted Effects Block button indicates that a particular effect is active within the current program. Pressing it once bypasses the effect; pressing and holding it causes the name of the effect to appear on the display. Similarly, pressing an Effect Block button after pressing the Edit button displays the name of the effect and allows you to edit its parameters.

The quality of the individual effects is generally outstanding. The reverbs are complex and transparent (though reverse reverb is conspicuously absent), the delays crisp and clean, the choruses

For manufacturer information, please see Contact Sheet, p. 163.

Inspiration...

motor mix™

Creative control at your fingertips! Motor Mix™ by CM Automation is the world's first dedicated worksurface for mixing digital audio. It is capable of operating any DAW software on any computer platform. Slightly larger than a sheet of notebook paper, Motor Mix has controls that are laid out like a traditional mixing console channel strip. Operating this worksurface is quick, intuitive and easy. You are in complete control of your digital audio mixing software. When inspiration hits, let the music be first . . . not the mouse.

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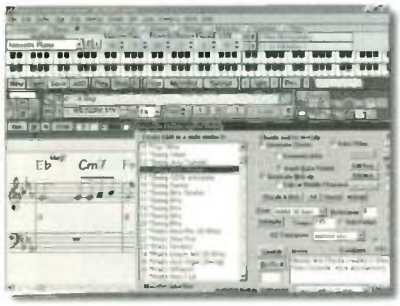

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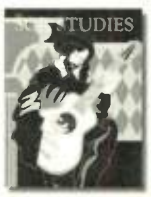

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THE PIANIST
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Stand Alone programs (or when Preamp is selected in the Gain Block), the other Gain "effects" are unavailable. The Preamp selection can produce overdrive and distortion sounds, but there's no way to switch between those tones and a clean setting without changing programs.

I also compared some of the digital emulations in the DSP Effects Blocks. The original Uni-Vibe had "Chorus" and "Vibrato" settings (though it was actually a phase shifter—and a very noisy, tone-sucking one at that). The G2's UniVybe emulation does a decent job of mimicking the original's Chorus setting, and with considerably less noise. (Another emulation, called CustomVybe, is a variation on UniVybe.)

Here's how several of the G2's other emulations hold up to the original pedals: OrangePhase captures the distinctive swoosh of the MXR Phase 90 with nearly dead-on accuracy. Red Comp closely matches a vintage MXR DynaComp, with similar attack and release characteristics, though it is not quite as bright. PedalWah 1 can be switched between Model C and Model V, which are emulations of Cry Baby and Vox wah-wah pedals, respectively. When used with the MPX R1's footpedal continuous controller, both versions sound quite good.

Finally, SweepFilter takes on the ambitious task of emulating a Moog resonant lowpass filter. I couldn't locate the '70s version of this unit, but I did compare the emulation to Big Briar's moogerfooger lowpass filter (designed by Bob Moog). SweepFilter wasn't as satisfying as the Big Briar unit, but in all fairness, nothing compares to the classic Moog-designed lowpass filter.

Even though the MPX G2's pedal emulations are not all dead ringers for the originals, it's important to keep in mind that the unit—especially when used with the R1 pedalboard—offers features that you wouldn't get even if you owned all of the pedals. Even the most elaborate custom-made switching system would likely fall short of the extensive routing and switching capabilities—not to mention MIDI control—provided by the powerful combination of the G2 and R1.

▼

Screamer is crunchier and more aggressive than the original.

OTHER OBSERVATIONS

The MPX G2 has many other features, too, including Reverb, Noise Gate, Master Bypass, and Speaker Simulator. Each of these functions can be set to Global status, making them immune to program changes. You can use the mono Speaker Simulator independently, Tap Tempo can be assigned to any parameter (not just to delay time), and MIDI Clock can be generated from the tapped tempo.

The G2's Preset Library is organized as a database, letting you sort presets using multiple criteria (number, guitar style, effect type, members of MIDI maps, and so on), though not all of them distinguish between application types, which limits their usefulness. The MPX G2 sends and receives MIDI Clock and Program Change messages and can be automated via SysEx. Also, most of its parameters can be adjusted in real time through Continuous Controller messages.

PROCESSING HEAVEN

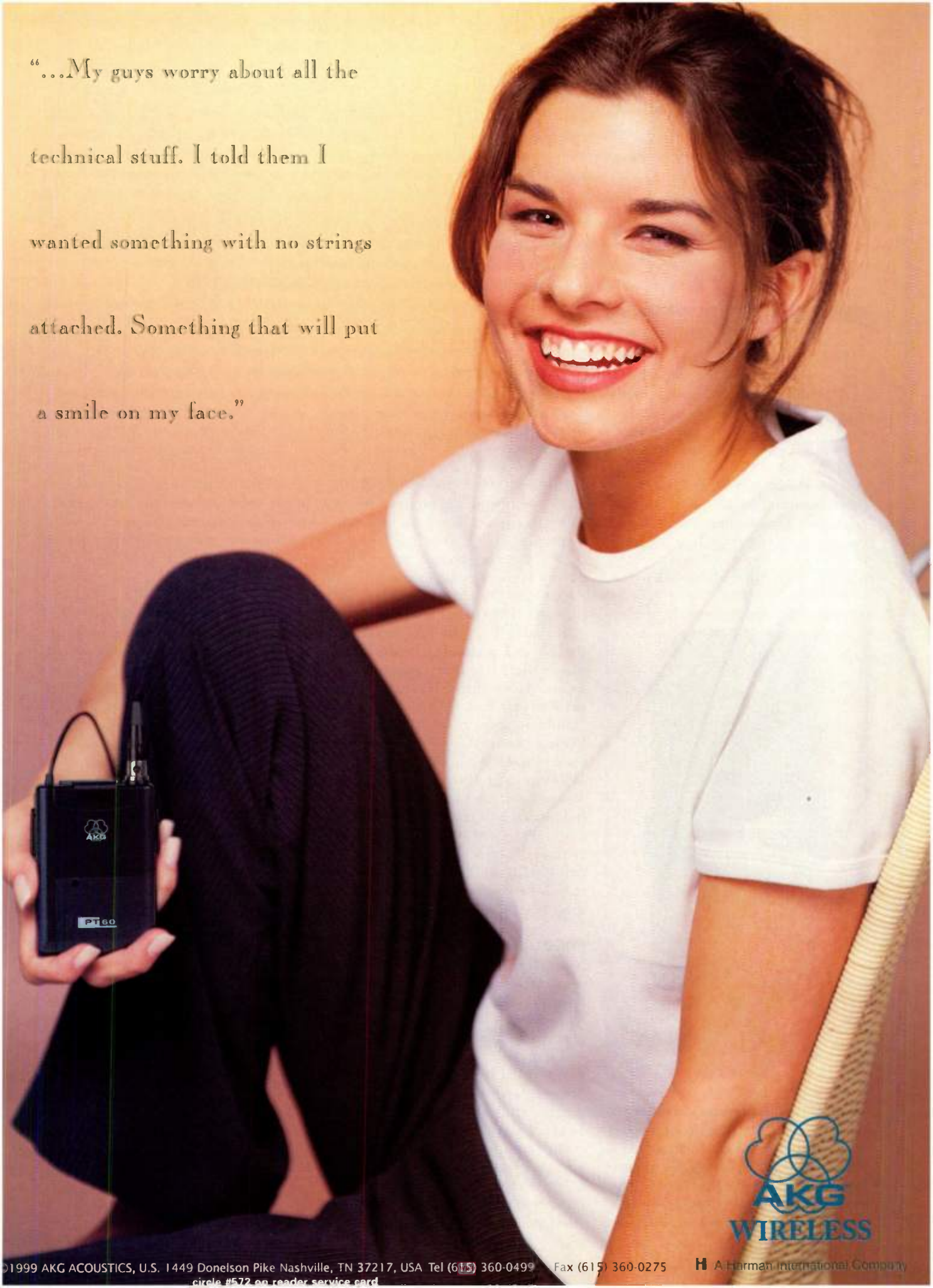
The MPX G2 is an extraordinarily flexible processor packed with terrific sounds and unique features that will be useful to guitarists of all stripes. A card slot for saving/loading programs—and perhaps stereo inputs for connecting the unit between a stereo preamp and a power amp—would have been nice touches, but on the whole I give the G2 an enthusiastic two thumbs up.

If you do purchase a G2, I suggest that you budget for the MPX R1 MIDI remote controller, too, as the two units are designed as separate parts of a complete package. Of course, at $1,499 for the G2 and $579 for the R1, this is by no means a bargain system. However, when you consider that a custom pedalboard and effects switching system costs several times as much—not including the pedals and rack effects—it is a bargain indeed.

Barry Cleveland *is the editor of* **EM**'s Personal Studio Buyer's Guide *when he's not composing, performing, recording, or playing guitar in the improvisational quintet Cloud Chamber.*

MPX G2 Specifications

Analog Instrument Inputs	(2) mono ¼" unbalanced with analog soft-clipping circuit and front-panel input-level control
Analog Outputs	(2) ¼" TRS balanced; (2) XLR balanced
Analog Return Inputs	(2) ¼" unbalanced with analog soft-clipping circuit and ganged level control on rear panel
Analog Send Output	(1) ¼" unbalanced
Other Ports	(1) ¼" TRS footpedal jack; (1) ¼" TRS footswitch jack; (1) ¼" headphone jack
MIDI Connectors	7-pin DIN for In/powered bidirectional remote; (2) 5-pin DIN for Thru and Out
ROM Presets/RAM Programs	250/50
A/D Converters (all)	24 bit
D/A Converters (all)	24 bit
Internal Processing	32 bit
Sample Rate	44.1 kHz
Frequency Response	20 Hz–20 kHz
Total Harmonic Distortion + Noise	<0.01% (@ 1 kHz nominal output level)
Dynamic Range (instrument input to send)	-110 dB (unweighted) minimum input to send with relay bypass on; -120 dB (unweighted) input to send with noise gate on; -97 dB (unweighted) typical instrument to main outputs in bypass
Dimensions	19" (W) x 1.75" (H) x 13" (D)
Weight	7.2 lbs.



NEUMANN

M 147 TUBE

The venerable company's most affordable tube mic to date.

By Myles Boisen

Any list of historic tube condenser microphones will necessarily include several Neumann models. Beginning with the CMV 3 "bottle" microphone in 1928 and continuing after the war years with the U 47 and U 67, among others, the company's development has encompassed most of the landmarks of tube condenser (not to mention solid-state condenser) technology. In short, when it comes to tube condenser microphones, Neumann is where it all began.

The M 147 Tube mic brings the sacrosanct Neumann tube sound within reach of the personal-studio buyer. Its supercardioid pattern and bountiful proximity effect make it harder to place than some mics, but when you position it just right on the desired source, the payoff is huge: you get a thick, luscious, near-magical sound.

In 1995, Neumann debuted its M 149 Tube, a condenser mic that addressed the current "tube mania" while setting a few standards of its own along the way. At $4,750, the M 149 has offered well-heeled enthusiasts an alternative to skyrocketing prices for vintage valves. But what about us working-studio stiffs who would like to have a big-name, large-diaphragm tube mic but can't afford going without food and shelter for the next few months in order to buy one? Neumann now has an answer: the M 147 Tube.

A REAL LOOKER

The M 147 is a fixed-pattern supercardioid studio condenser mic featuring the K47 dual-diaphragm capsule (the modern version of the legendary M7) within a nostalgically rounded mesh enclosure. Despite its retro look, the M 147 is thoroughly modern inside, with electronics based on the transformerless M 149 Tube microphone. In fact, these two contemporary transducers are so similar in their output circuitry that Neumann ships them with the same N 149 external power supply.

In the M 147, the K47 capsule is paired with a slender vacuum tube whose origin is a closely guarded trade secret. According to Neumann product manager Karl Winkler, the tube is "a subminiature triode similar to the [Telefunken] AC701k tube that was used in the KM 54, KM 64, M 49, and M 269 mics, but with better specs." Neumann doesn't make the tubes, but it does carefully test each one. "Out of a typical lot," says Winkler, "a large percentage are rejected," and he cites the painstaking selection process as one reason behind the M 147's impressive specs (see "M 147 Tube Specifications").

FREQUENCY RESPONSE

Neumann's published frequency-response chart shows a rising curve from 20 to 300 Hz, a flat response from 300 to 1,500 Hz, and gentle (3 dB) presence peaks centered at 4 and 10 kHz. The mic avoids excess sibilance with a slight dip at 6 kHz, between the two presence peaks. Gain at 100 Hz is roughly 5 dB down relative to the 4 and 10 kHz points.

The resulting character of the M 147 is generally airy, well defined, and just a little exaggerated in the high end; smooth to thick in the mids; and true (though not ruler-flat) in the low end. A significant factor not indicated on any chart is this mic's proximity effect, which is substantial. All directional microphones tend to increase bass frequencies of source sounds the closer they get to the source; in the M 147's case, bass boosting is noticeable up to one foot away and can contribute greatly to low-end and midrange "warmth."

ACCESSORIES

All metal parts on the microphone and accessories are machined with precision, maintaining an old-fashioned standard of quality that has become increasingly scarce in the corner-cutting climate of today's globalized mic market. My only gripe is with the 10-meter cable that connects the M 147's 8-pin output jack to the power supply. The cable is pretty spindly, and it kept getting kinked up during my test sessions—raising some concern that it may be prone to failure over time.

Heavy-duty threaded connectors on both ends of the cable are a nifty feature, however, as is the "smart" N 149 power supply, which will not provide tube-heater voltages unless the mic is properly hooked up. The supply can also be connected to phantom-powered preamps without damage.

I'm happy to report that the swivel mount that ships with the M 147 is a much-improved nylon-and-metal upgrade of the flimsy mount originally included with Neumann's TLM 103. Moreover, Winkler informed me that the final version of the swivel mount (model SG1) will be solid metal for both microphones and should be available by the time this review goes to press.

Other accessories for the M 147 include a detailed user manual and an oversized nylon mic bag with a Velcro seal. All accessories fit neatly in a sturdy aluminum flight case sporting classy black trim and a bold Neumann logo on one side. (Actually, the logo is a bit *too* bold for my tastes; I would prefer not to advertise the case's prized—and pricey—contents when carrying it in certain neighborhoods.)

As you might expect, the M 147 comes without a shock mount. There is one available, however: the $195 model EA1.

TESTING 1, 2, 3

To get acquainted with the Neumann M 147, I first performed some comparison tests with three other modern, large-diaphragm tube mics from the

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the process. On a music box, the mic's low noise floor and full-bodied sound beat out the Manley tube mic, which sounded too "pingy" by comparison. The M 147 also provided an excellent old-time tone on a bowed saw, with a generous low end—even at a distance of two feet—that helped offset unwanted bow scraping.

In a jazz ensemble, I positioned the M 147 to pick up a blend of direct sound and reflections from the side of a French horn's bell. That orientation contributed to more leakage than usual from the other instruments, but the M 147 still captured a rich airiness from this murky instrument, and the off-axis sound was quite acceptable.

During a multiple-mic comparison on clarinet, reed player David Alt voiced a preference for the sound of the M 147 over that of the other tube mics. Alt also noted that on his alto and tenor saxophones, the M 147 gave a good balance of highs and lows with "lots of clarity;" he felt, however, that its tube coloration sounded "too processed" and that the low notes seemed muffled.

VOCALS

There's nothing quite like a tube mic for male vocals, and contrasting Neumann's new tube mic to the Lawson L47MP proved very revealing. On folksy, untrained, or simply mediocre voices, the Lawson was usually smoother and more forgiving. On a singer with a creamy, Johnny Mathis–like voice, the two mics produced nearly identical timbres, except that the M 147 stamped its incisive presence trademark on the track. With a more strident male vocalist, though, the M 147 sounded decidedly scratchier around 5 to 7 kHz and less full in the low end than the Lawson mic.

Just the same, I tracked with the M 147, reasoning that the song we were working on (a sassy Stones-ish number) could use some extra "gravel" from the vocals. We got a wonderfully exciting and edgy tone that jumped right out of the track without ever sounding too harsh or biting.

This extraordinary quality, which I once heard described as "tube splatter," is characteristic of the best vintage vocal mics, and our track had a double dose of it. A technical explanation would likely attribute this tonal quality to natural compression and the warm sound of harmonics generated in the tube. If you'd prefer an indulgence of adjectives, word slingers in the control room came up with a few that might more commonly be heard at the beach during a summer scorcher: hot, wet, sandy, and fat!

The M 147 was absolutely magical on a nylon-string guitar.

My studio partner, Bart Thurber, reported that the M 147 worked out very well for him as a vocal mic on both female and male rock singers. He also commented that any roughness or congestion in a voice was often highlighted by the M 147's characteristic presence boost.

The M 147 can be relied on to thicken any lead vocal, particularly with female singers. And yet, for all its virtues, this mic does tend to overemphasize treble frequencies above 3 to 4 kHz on certain instrumental sources. For example, it couldn't match the even upper mids of the Lawson L47MP on a tenor-sax solo; and compared with a Coles 4038 ribbon mic on cello, it sounded much too scratchy. And despite my enthusiastic urging, another engineer declined to use the M 147 for a traditional jazz kick drum, opting instead for a dynamic mic that captured a less defined attack.

SMALL WONDER

In addition to their celebrated status as "must-have" mics for recording vocals, tube condensers can also work wonders for amplifiers of all types, as well as for drums, strings, horns, and other acoustic instruments. As Neumann's newest entry, the M 147 Tube is certainly up to all these tasks and many more. When you factor in the advantages of modern low-noise electronics, quality control, reliability, and a (relatively) affordable price, the M 147 looks very good next to its costly cathode cousins.

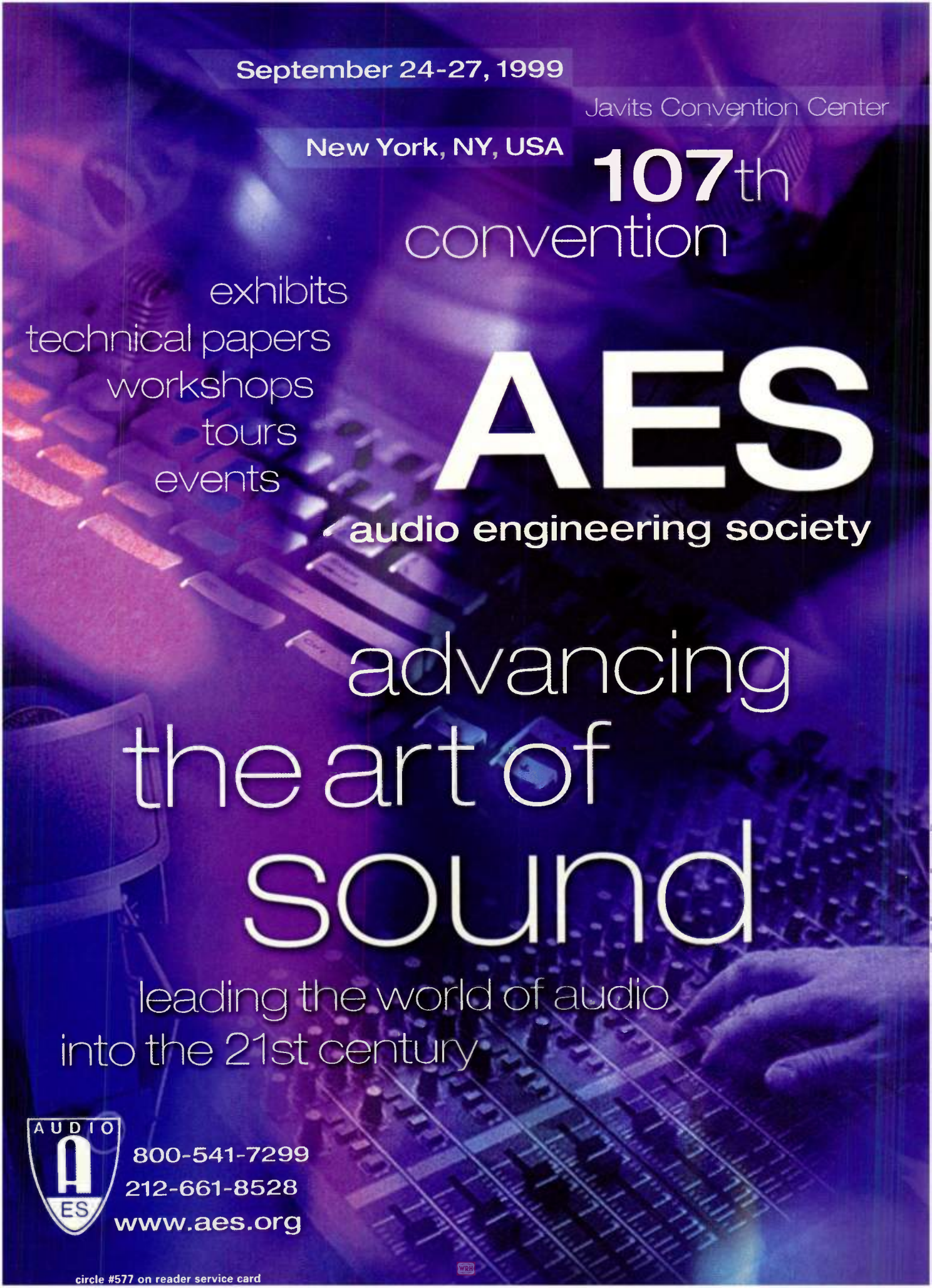
As is the case with many premium mics, careful listening and positioning are the prerequisites that make spectacular results possible. The M 147's only serious drawback is a tendency to sound slightly raspy or bass-lean in some applications—especially when distant placement negates the proximity effect. (Then again, place this mic too close to the same source and it may yield an overly dark, bottom-heavy sound.) On the other hand, the M 147's signature "edge," centered at 4 kHz, is just what many instruments need to cut through a mix with little or no extra equalization.

I highly recommend the M 147 for rock, rap, pop, jazz, and blues vocals; drum-room or kick-drum miking; all tube and solid-state instrument amps; nylon string guitar; low-volume or indistinct sound sources that need some extra presence; and any type of digital recording. In short, I like the M 147 a lot—so much so that I bought one.

Myles Boisen *is a guitarist, producer, composer, and head engineer/instructor at Guerrilla Recording and the Headless Buddha Mastering Lab in Oakland, California.*

M 147 Tube Specifications

Acoustic Operating Principle	pressure-gradient transducer
Diaphragm	1" gold-vapor-deposited Mylar
Polar Pattern	supercardioid
Frequency Response	20 Hz–20 kHz
Dynamic Range for THD <0.5%	100 dB
Dynamic Range for THD <5%	120 dB
Signal-to-Noise Ratio	80 dB
Self-Noise	12 dB (A weighted)
Sensitivity	20 mV/Pa (@ 1 kHz)
Maximum SPL for THD <0.5%	114 dB
Maximum SPL for THD <5%	134 dB
Dimensions	5.59" (L) x 2.24" (D)
Weight	1.01 lbs.



LYNX STUDIO

LYNXONE (WIN)

With this card in your computer, you're really wired for sound.

By Zack Price

If you're in the market for a high-quality 24-bit audio mastering card with support for 96 kHz digital I/O (48 kHz through its analog inputs), as well as word-clock and MIDI I/O, then Lynx Studio Technology's LynxOne deserves a close look. This half-size PCI card boasts several professional features and still boasts a very affordable price tag.

The LynxOne's four analog and two digital audio cables connect to the card through a 25-pin D-sub connector. Each cable is six feet long, which I found sufficient for connecting all of my digital and analog devices. If you need to extend the length, you can use additional cables.

The digital cables use standard AES/EBU (XLR-type) connectors. To interface with S/PDIF electrical connections, you'll need to buy or make properly wired AES/EBU-to-S/PDIF adapter cables. (The LynxOne manual, available at Lynx Studio Technology's Web site in Adobe Acrobat format, provides information on how to configure these cables.)

I appreciate that the analog audio cables likewise use XLR-type connections because balanced lines tend to reject noise better than unbalanced lines. The inputs and outputs offer a generous 16 dB of headroom to accommodate signals with wide dynamic swings. Lynx recommends that you connect the card's line-level outputs to the line-level inputs on external equipment, instead of using the XLR-type microphone inputs available on many mixers. Doing so will help you avoid encountering distorted signals caused by an overdriven microphone preamp.

The LynxOne also includes a pair of standard MIDI In and Out jacks as well as word-clock in and out jacks with BNC-type connectors. These female jacks are provided on cables that attach to the card with a 15-pin D-sub connector. Each cable is two feet long, which is more than adequate to ensure that the cables can be routed away from any other card connectors that might be protruding from the back of your computer.

LynxOne
Minimum System Requirements
90 MHz Pentium or Alpha processor; 16 MB RAM; Windows 95/98/NT; PCI slot

You get 32 channels of MIDI I/O on the MIDI jacks, which can also transmit or receive MIDI Time Code for synchronizing other devices and applications with the LynxOne. Similarly, you can use the word-clock jacks to synchronize the LynxOne with other devices that support either standard or 256-word clock. The LynxOne also synchronizes to 13.5 and 27 MHz video clock signals.

The LynxOne has internal header connections for synchronizing to other PC cards with clock input and output, such as the DPS Perception video card. This should come as no surprise because the founders of Lynx Studio Technology formerly worked for Antex Electronics, a company that manufactures audio cards for audio-visual professionals.

INSTALLATION

The LynxOne comes with drivers for Windows 95/98 and Windows NT 4.0 for Intel and DEC Alpha processors. The company should be shipping DirectSound drivers by the time this issue hits the newsstands; ASIO drivers are expected to follow shortly thereafter. Configuring the card and the drivers is simple. First, install the drivers into your operating system—a process that happens so quickly, all you may see is a brief blip on the screen. Next, reboot your computer, and Windows should recognize the new hardware automatically.

Speaking of hardware installation, you can install up to four LynxOne cards in your computer for a total of 16 audio and 128 MIDI channels. Moreover, all four cards share the same IRQ, which saves system resources. (If you need that many channels of audio I/O, though, I suggest getting a multichannel card instead.)

You can connect and synchronize multiple LynxOne cards using the header jacks found on each card. For example, connect the master card's header output (which can be any card

Lynx Studio Technology's LynxOne audio card is an excellent choice for mastering. With support for digital I/O at rates of up to 96 kHz at 24-bit resolution, A/D/A converters that offer 48 kHz and 24-bit support, 32 MIDI channels, and word clock, the LynxOne is suitable for today's professional studio.

Somebody's Been Thinking About You

In designing our new workstation we had to carefully consider many things, but above all, the most important was what you might actually want. So if you're about to settle for the usual mix of synth and sequencer with a few extra frills thrown in; wait ! Wouldn't you like to have a top of the line pro piano too? What about a drawbar organ? Or how about a universal sample player? Or a groove machine? The Equinox Pro is all of these things and much, much more at an unbelievable price that makes sweet dreams come true. Take a look.

SOUND LIBRARY - Nearly 1200 sounds "out of the box", ranging from the phattest analog synth patches to crisp acoustic and orchestral tones, and without having to buy any additional ROM modules. These sounds, created by some of the top programmers in the U.S, can be quickly and easily edited and over 2000 user sounds can be saved into the standard on-board memory.

THE REAL PIANO* - Generalmusic's PRO series physical modeling pianos hardly need any introduction at all. An established favorite of many of the world's most prominent recording and touring pianists, the Grand Piano, Rhodes, Wurli, CP and Clavinet sounds from the, "top of the line" PRO2, are included as standard on the Equinox PRO along with our patented "Damper physical model" and "Natural string resonance" technologies.

PERFORMANCE - Combine up to 16 sounds together in any kind of multi-split or layer configuration. The sliders and buttons can instantly be used as mutes and volume controls for your 16 parts.

DRAWBARS - One innocent looking little button transforms the Equinox PRO into a versatile drawbar organ. While the sliders become a set of traditional organ drawbars, the front panel offers instant access to Key Click, Percussion and Rotary speaker speed, (all user editable). There's even a choice of Drawbar voicings, (Smooth, Hard, Jazz and Rock) and the added flexibility to edit each drawbar's pitch and pan position individually.

REAL TIME CONTROLLERS - Whether you're playing single sounds, performances, samples, grooves or sequencer songs, grabbing a handful of sliders will instantly transport you to analog heaven, (and without the need for WD40). Sliders are pre-assigned to control envelope attack/ decay/ release, filter cut-off / resonance and LFO depth and speed but any slider can easily be re-programmed to control the parameter of your choice. In sequencer mode the sliders can also be set to function as a 16 track mixer with the press of a single button.

DRUMS KITS - 43 on board drum kits from a brand new sample library ranging from hip-hop, dance and techno to fusion, rock, jazz and 70s. Editing is quick and easy and you can save as many user drum kits as you like, (up to 2048 kits).

USER PANEL - Each of the 16 sliders and 16 buttons, (8 physical buttons and sliders with a SHIFT key), can be individually programmed to control either the internal functions of the synth engine and sequencer or they can be set up to send any MIDI controller, system common or system real-time message of your choice to either of the two MIDI out ports. Up to 16 complete panel configurations can be saved and independently recalled.

4 INDEPENDENTLY ASSIGNABLE OUTPUTS - Each performance part, sequencer track or drumkit instrument can be individually assigned to any of these outputs either as a stereo pair or as a single mono out.

POWERFUL SEQUENCER - The versatile 16 track sequencer offers a staggering 250,000 events of storage and allows you to store 16 songs in memory at once. Add to this the power of 1/192 resolution, Groove quantize, Event list editing and the ability to record ALL slider movements, (either real-time editing of filters, envelopes etc or multiple track mixdowns) plus the life saving UNDO function and you'll start to see why the Equinox PRO is really a composer's dream machine. The sequencer even has it's own 16 tracks, independent from the rest of the synthesizer so, while a 16 track song is playing, you can freely select any complex multi-split or layered performance to play along with it. Simply put, it's just like having two separate keyboards.

EFFECTS - Up to four independently assignable DSPs are available offering 85 crystal clear digital effects. Ranging from simple reverbs and choruses to complex composite algorithms like GUITAR FX, (Distortion, Gate, Delay and 4 Band EQ using only one DSP section !), 3D ENHANCER, AUDIO EXCITER, RING MODULATOR, 10 BAND EQ and 4 PART PITCH SHIFTER.

SINGLE INSTRUMENT GROOVES - Any musician will find our collection of "single instrument grooves" indispensable. These superb phrases, offering everything from drum loops and bass lines to clavinet riffs, organ licks, guitar grooves and a whole lot more, have been created by some of the worlds top session musicians. And they're yours to use as you please in any sequence or song creation of your own. We've already taken care of the royalty payments !

CONTROLLER/6 MIDI PORTS - With no less than 6 independent MIDI ports, the Equinox PRO offers full control over 32 channels of MIDI data. The keyboard can be used as a 16 zone controller with each zone having its own programmable upper and lower split points and velocity limits, (which may overlap with other zones - of course !).

3 PROGRAMMABLE PEDAL JACKS

ARPEGGIATOR - With 16 factory presets and 16 user programmable patterns the Equinox PRO's Arpeggiator will send you soaring back to the 70's, (with a few little technological miracles thrown in for good measure). As well as the usual UP and DOWN directions there's also RANDOM and INPUT, (the order in which the notes were played). Velocity can be pre-set, disabled or controlled by your playing style and there's also a CRESCENDO feature to reduce or increase velocities as the pattern cycles around. There's even a HOLD button which lets your pattern do it's thing while giving you both hands free to tweak the arpeggiating sound with the real time controllers.

DJ GROOVES - Having 1024 on-board grooves puts the Equinox PRO head and shoulders above any of those stand alone "Groove Stations". All grooves can be edited, filtered, remixed and even "scratched DJ style" in real time. Plus you can create your own grooves from any midi file or sequencer song.

HARD DISK* - The Equinox PRO comes as standard with a 2Gigabyte internal hard disk. Anything can be stored on the hard drive - Sounds, Performances, Drum Kits, Imported Samples, Grooves, Midi Files etc. The hard disk can also be accessed "invisibly" so that new songs can be loaded while the sequencer is already busy playing another song.

SCSI INTERFACE*

88 NOTE PIANO ACTION - Fully weighted 88 key hammer action gives you the authentic feel and response of a fine acoustic piano.

SAMPLE TRANSLATOR - The Equinox PRO comes standard with 8Mbytes of Sample RAM, (expandable to 40Mbytes), which can be used to load samples in Akai, Kurzweil, WAV and AIFF formats from CD-ROMS or floppy disks, (with other formats to be supported in future o.s releases). Loaded samples can be freely edited like any other sound and they even stay in memory after you switch off the power.

EQUINOX PRO

Breakthrough Price: $ 3,695.00

Also available: Equinox 61 $ 1,995.00
Equinox 76 $ 2,195.00

* Optional on Equinox 61 & 76. models.

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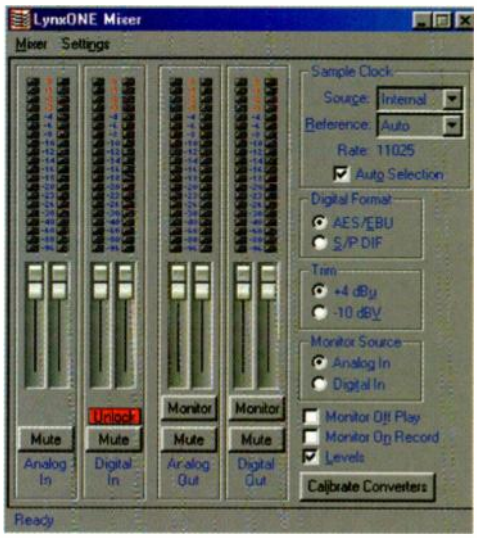

FIG. 1: The LynxOne Mixer window lets users select Sample Clock settings and reference sources, choose digital audio formats, and determine whether analog I/O will be balanced or unbalanced. In addition, the LynxOne Mixer allows you to periodically recalibrate the card's digital audio converters as needed.

you choose) to the slave card's header input. If there are additional cards in the system, connect the header output of the second card to the header input of the third card, and so on. You can achieve the same results using the external word-clock connectors, although things might get a little tangled up behind the computer as a result.

MIXER MIXER

The LynxOne Mixer software controls the various aspects of the card's functions (see **Fig. 1**). For instance, you can adjust the volume for the analog and digital inputs and outputs. For optimum performance, Lynx Studio Technology recommends that you leave these settings at their maximum; that way the card (and by extension, the computer and the software) won't have to perform any more calculations on the digital audio data than necessary.

You can select the Sample Clock source settings (Internal, External, Header, and Digital) from the LynxOne Mixer. In Internal mode, the sample clock is derived from the card's own crystal, whereas the Digital setting causes the LynxOne to receive its sample clock from the Digital Audio In jack. The External sample clock source comes from the external BNC connector; the Header clock source comes from a device connected to the card's Header jack. The clock reference choices are the video and word-clock settings that I mentioned previously.

The Mixer is also where you choose between AES/EBU and S/PDIF digital format settings and switch between +4 dBu and -10 dBV settings for analog input and output. From there you

LynxOne Specifications

Audio I/O	2/2, balanced or unbalanced, on XLR breakout connectors
Digital I/O	1/1, AES/EBU or S/PDIF format, on XLR breakout connectors
Maximum Digital Transfer	24 bit
Sampling Rates via Digital Connectors	32, 44.1, 48, 88.2, 96 kHz
Sampling Rates via Analog Connectors	any rate between 8 and 48 kHz
A/D and D/A Converters	24 bit, 128x oversampling, delta-sigma
Synchronization Types	MIDI, word clock, Superclock, 13.5 and 27 MHz video clocks
Bus Type	PCI
MIDI Ports	(2) In/Out; 32 channels
Input Total Harmonic Distortion	0.0022% typ., 1 kHz signal @ -1 dBFS, 22 Hz–22 kHz (analog input to digital output)
Output Total Harmonic Distortion	0.0015% typ., 1 kHz signal @ -1 dBFS, 22 Hz–22 kHz (digital input to analog output)
Additional Features	internal header connector for sync with other cards; software mixer application

For manufacturer information, please see Contact Sheet, p. 163.

can also elect to monitor the analog or the digital inputs. The selected monitor source is mixed with the playback audio on the analog and digital outputs using the LynxOne's low-latency mix engine. (As with most audio cards, you can't monitor both the analog and digital inputs at once.)

Another important Mixer feature is the Calibrate Converters button, which calibrates the DC offset of the A/D and D/A converters. When the LynxOne driver first loads, it automatically calibrates the converters. However, temperature changes within the computer over time may cause the DC offset to drift. Lynx Studio Technology recommends that you recalibrate the converters after your computer has been on for 15 to 30 minutes, or whenever any sudden changes in room temperature occur.

MASTER CARD

The LynxOne is an excellent mastering card in terms of sound quality and flexibility. In fact, I mixed some digital audio sequencer tracks from my main computer to a second computer running the LynxOne card and a multitrack hard-disk recording program. I transferred the digital audio tracks from the main computer using the digital connections between the main computer's card and the LynxOne. The LynxOne's analog inputs were connected to my outboard mixer, which combines all of my MIDI gear into a stereo output. By using the LynxOne's analog and digital inputs simultaneously, I was able to create a four-channel premaster that I subsequently mixed down to a final stereo mix.

The LynxOne would work well as a four-channel audio card for the personal-studio owner. The 32 channels of MIDI I/O can also serve as a useful adjunct for users of digital audio sequencers and other MIDI applications. Additionally, the card will likely find a significant market among audio-visual professionals, given the features that it offers and the background of the company founders.

The LynxOne's only drawback is that it has no optical S/PDIF connections. Many musicians can probably live without optical connections, so I don't regard this as a major deficiency. Besides, you can always purchase an inexpensive electrical-to-optical S/PDIF converter box, if necessary.

All told, the LynxOne is a great-sounding, solid performer and a worthy addition to any studio.

Zack Price *is an audio editor and Windows digital audio consultant in the Chicago area.*

LYNX STUDIO TECHNOLOGY
LynxOne audio card
$549

FEATURES ■■■■
EASE OF USE ■■■■
AUDIO QUALITY ■■■■■
VALUE ■■■■■
1 2 3 4 5

PROS: Excellent sound. Well constructed. Balanced audio connectors. Multiple synchronization and MIDI I/O options.
CONS: No optical S/PDIF connections.
CIRCLE #442 ON READER SERVICE CARD

A.R.T.

DMV-PRO

A versatile multi-effects processor with four discrete channels.

By Jeff Obee

There has been a dramatic shift in the realm of multi-effects processors over the last few years: prices have fallen considerably, audio quality has improved, and true-stereo, multiple-I/O processors have become the norm rather than the exception. The A.R.T. DMV-Pro fits this description nicely. It features four independent, 24-bit DSP engines and four discrete inputs and outputs with 20-bit converters, and you can choose from the unit's four routing configurations with the touch of a button.

PRO OVERVIEW

The DMV-Pro's four DSP engines are routed through two processors (named A and B) using A.R.T.'s proprietary Dynamic Engine Allocation software, which manages and maximizes the unit's resources. This gives you peak audio performance regardless of how many DSP engines are in use. And each processing engine responds to a full range of programmable MIDI control data.

Effects include Chamber, Room, Hall, Plate, Delay, Pitch, Chorus, Flanger, Phaser, Rotary, Tremolo, and Panner, with six variations of each. All the effects algorithms take full advantage of the DMV-Pro's processing power.

A stylish, turn-of-the-millennium design defines the DMV-Pro's front panel. There are three multifunction data wheels (referred to as *encoders*), a power button, two edit buttons, and four global-parameter buttons (Routing, MIDI/Utility, Store, and Bypass). The displays include three LCD screens, two input meters (one for each for processor), and four LEDs for the routing configurations.

The back panel of the DMV-Pro has four unbalanced ¼-inch inputs, four unbalanced ¼-inch outputs, MIDI In and Out (no Thru), and level switches for both the inputs and outputs that set them to either instrument or line level (see **Fig. 1**). There is no digital I/O on the DMV-Pro.

INTUITIVE INTERFACE

The first thing I do when I get a new effects processor is see how quickly I can dial in an effect. So as always, I racked up the DMV-Pro and began programming. The logic of the DMV-Pro's front-panel layout was immediately apparent, allowing me to get the results I wanted instantly before ever referring to the manual.

The multifunction encoders can be used to select a program, audition edit parameters, access Smart Encoder Mode, or load a program. Smart Encoder Mode lets you use the encoder to tap in delay times, audition reverbs, or toggle between high and low rotary-spin rates. Aside from the very basic edit and global-parameter access buttons, the system of dialing and pressing the encoders is the only way to input data on the DMV-Pro—there are no alphanumeric keypads.

ROUTING AND GLOBAL

The effects-bus matrix provides four routing options: Twin Stereo, Cascade, Multi FX, and Discrete 4. Twin Stereo gives you two separate stereo processors, with inputs 1 and 2 feeding DSP engines 1 and 2, and inputs 3 and 4 feeding DSP engines 3 and 4. In this configuration, modulation parameters are linked across both stereo channels, but in the reverb algorithms the parameters remain in their distinct audio channels to accurately emulate acoustic environments.

In Cascade mode, inputs 1 and 2 feed the two stereo engines in series so you can stack stereo effects. (Inputs 3 and 4 are inoperative in Cascade mode.) Multi FX sums inputs 1 and 2 to mono and then runs them in series through two mono engines in processor A. The signal is then split and fed into stereo processor B.

In Discrete Mode, the DMV-Pro acts as four independent mono processors, with each input feeding its corresponding output when all jacks are used. Input 1 takes priority and must be plugged in. If input 1 is the only jack used, DSP engines 2, 3, and 4 will all receive the signal from that input. If inputs 1 and 3 are plugged in, engines 1 and 2 are fed from input 1, and engines 3 and 4 are fed from input 3.

A Global Mix function sets the wet/dry mix for every effect on the unit and overrides any settings made in each individual effect. A feature called Dry Kill globally removes any unprocessed signal in the mix bus (that is, it automatically makes each effect 100 percent "wet"), optimizing the DMV-Pro for use with a mixing console.

EFFECTUAL MUSINGS

Each of the 100 factory presets is fully editable. The first 19 are a "best of" collection, followed by small groupings of each effect type. The DMV-Pro has a wide assortment of useful presets, but these only scratch the surface of the unit's potential. I used them as a starting place to familiarize myself with the unit's different features, then moved on to custom settings. Fortunately, the effects are very easy to customize.

To load an effect, turn the encoder to the preset you want and press it. To begin editing, press the Edit button. Once you have entered Edit mode, you can select one of six algorithmic variations (referred to as "classes" on the DMV-Pro) of the basic effect. Some classes are available in every effect, while others are effect specific. For example, in a Hall reverb you can choose from Vocal, Instrument, Concert, Gated, Ambient, or Dynamic classes. In the Phaser program, you will find 5-Notch Ducking, MXR Phase 90, MXR Phase 100, Gated, Boston Phaser, and Dynamic classes.

The Boston effects class is notable because it can process either the difference information alone

The DMV-Pro has a sleek user interface featuring three multipurpose data wheels, three LCD screens, and an input meter for each of the processors.

Drums

Times Infinity

And then some...

The Alesis **DM Pro**™ holds 1,664 of the best acoustic and electronic kicks, snares, toms, cymbals, hi-hats, percussion, chromatic/tonal sounds, special effects and much more. For MIDI studios, pro recording facilities and performing percussionists, DM Pro has 64 programmable kits with up to 64 instruments in each. For triggering from pads and re-triggering from tape, you get 16 lightning-fast trigger inputs and 6 independent audio outputs.

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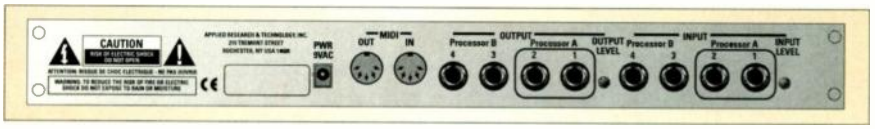

FIG. 1: The back panel of the DMV-Pro is as uncomplicated as the front panel, with an input- and output-level button, four discrete unbalanced ¼-inch inputs and outputs, and MIDI In and Out ports.

(those parts of a mix that do not occur equally in both channels) or the full mix, depending on the setting of a parameter called *difference factor*. For example, processing the difference information would allow you to flange just the room ambience and not the foreground instruments. (Boston effects work the best on stereo-miked instruments.)

The DMV-Pro's reverbs are just right for creating quality, custom environments without having too many parameters to bog you down. Of note is the Motion parameter, which lends subtle movement to reverb tails as they decay through the cross-coupled stereo field. The DMV-Pro's reverbs measured up beautifully under a variety of everyday applications, including vocals, drums, and percussion.

The maximum number of memory locations is 100. Presets are numbered rather than named: unless you have a good memory for numbers, be prepared to keep a list of your favorite patches so you don't inadvertently overwrite them. If you edit an effect or build one from scratch, the DMV-Pro will remind you that saving your new effect will erase the current preset. In the event of an emergency, you can reinitialize the unit using the Factory Reset routine, but in doing so you will lose all your custom settings.

MODULATIONS

Within each of the modulation effects—Chorus, Phaser, Flange, Tremolo, and Panner—you can choose from a palette of LFO shapes: Exponential, Triangle, Sine, Rectified Sine, Inverted Rectified Sine, Square, Ramp, and Inverted Ramp. You can also offset the LFOs in 30-degree increments up to 360 degrees, enabling you to modulate one side of your stereo sound differently from the other.

The DMV-Pro's Chorus effects are smooth and clean, with a pleasant stereo spread. I like the swelling effects that can be created using the MXR Ducking class and setting the threshold to trigger below a certain level. I was able to create some lustrous washes by chaining a chorus with a reverb in Cascade Mode.

I tend to think of a phase shifter as a low-tech-sounding device. But having a clean, versatile, 24-bit model is a real joy. The DMV-Pro's Phaser algorithms lend a wonderful presence to synth patches and work wonders on drum loops. The flangers are similarly amazing, running the gamut from traditional guitar-oriented sounds to the seriously tweaked.

DELAY AND PITCH

Each channel features a digital delay of up to 1.25 seconds; you can, however,

DMV-Pro Specifications

Inputs	(4) ¼" unbalanced
Outputs	(4) ¼" unbalanced
MIDI Connectors	In, Out
Memory Locations	100
A/D/A Conversion	20 bit
Internal Processing	24 bit
Sampling Rate	46.875 kHz
Frequency Response	10 Hz–20 kHz (+0/-2 dB)
Dynamic Range	>93 dB
Total Harmonic Distortion	<0.01% @ 1 kHz
Power	9 VAC (wall wart)
Dimensions	19" (W) x 1.75" (H) x 6.5" (D)
Weight	5 lbs.

For manufacturer information, please see Contact Sheet, p. 163.

A.R.T.
DMV-Pro multi-effects processor
$499

FEATURES ■■■■
EASE OF USE ■■■■■
AUDIO QUALITY ■■■■■
VALUE ■■■■
1 2 3 4 5

PROS: Excellent sound quality. Easy to use.
CONS: No MIDI Clock sync. No digital I/O. Only 100 memory locations. Presets cannot be named.

CIRCLE #443 ON READER SERVICE CARD

attain up to five seconds of delay by chaining together the mono delays. There are a number of delay parameters that make for exciting textural processing. One is the Mic Placement class, which utilizes a specialized multitap delay system to create stereo imaging. This class simulates close and far miking of a speaker cabinet as well as different speaker-cabinet sizes. The DMV-Pro lets you determine mic distance (from the speaker cabinet), cabinet size, mic separation (distance between mics), and mic position (vertical and horizontal).

In the other delay classes, you can choose up to 12 delay taps, and there is a collection of 18 preset tap shapes to bring your delay patch to life. The Ducking class enables you to set a threshold where the delay is triggered, along with ganged attack/release times. It would have been great to have a tap-tempo button on the front panel. Instead, you have to choose Tap in Smart Encoder Mode and then hold the edit button while tapping the encoder dial, causing the dial to spin through parameters in the process. You can also set tap times via MIDI. One other interesting and useful delay feature in Smart Encoder Mode is the Repeat Hold, which allows you to capture and loop a sound indefinitely.

The DMV-Pro's pitch shifting algorithms sound good and are quite stable for a processor in this price range. But when you use a MIDI expression controller with it, don't expect to get zipperless pitch transposition—that's asking too much However, I was able to get some weird yet pleasing effects from the DMV-Pro, dressing up some of my old Kurzweil patches in ways that enhanced them enormously. For example, using delays in the 75 ms range, lots of regeneration, and low-mix values, I was able to create pitch arpeggiations that were subtle and effective.

There is a very cool dynamic-sensing algorithm in the Pitch class called Whammit. When the threshold you've set is crossed, portamento is triggered, causing the pitch to glide up or down at a user-selected rate to the desired transposition. The Micro 2-Voice class provides two pitch shifters with a range of 1.27 semitones, up or down. The Full 2-Voice goes up a maximum of 12.7 semitones and down two octaves. On both the Micro 2-Voice and Full 2-Voice, the transposed channels go to different outputs.

MIDI, MIDI, ON THE WALL

You can assign MIDI data within an effect patch or globally. These operations are easy because of the simplified menu and the clear way in which the unit is laid out. Global MIDI is accessed through the MIDI/Util button, allowing you to set the base MIDI channel (1 through 16, or Omni), perform SysEx dumps, and assign MIDI Mappings. The MIDI Dump function performs one task: transmitting all presets. The DMV-Pro will respond to dumps of individual effects or other SysEx functions, but ignores System Reset and Active Sensing messages.

I was very disappointed to discover that you cannot sync to MIDI Clock. This is a severe oversight. In my opinion, any modern effects processor should offer that capability. With the DMV-Pro, you'll have to rely on math and your ear for delay and LFO sync.

FOUR THUMBS UP

The DMV-Pro is clearly a winner. It sounds fabulous, it's easy to use, and it's a great deal, offering two stereo effects processors in one unit at a very reasonable price. The routing capabilities of the DMV-Pro give you a wide variety of processing options. The only major gripe I have is that there's no MIDI Clock sync. Other than that, the DMV-Pro is a pleasure to have in the rack.

Jeff Obee *is a fretless bassist, synthesist, and composer. Reach him at obeej@dsp.com.*

D U Y

DSPIDER 1.1 (MAC)

The first TDM plug-in for creating original plug-ins.

By Peter Freeman

DUY first appeared on the Pro Tools radar in 1996 with a highly regarded suite of TDM plug-ins: *Shape, Valve, Wide, Tape,* and *Max.* (*Shape, Tape,* and *Valve* were reviewed in the March 1998 **EM**.) These products represented something of a departure in the TDM world, where the landscape was overpopulated with workaday plug-ins such as EQs and dynamics processors.

Although these programs have been updated, no new products followed until the Barcelona-based company released the cryptically named *DSPider.* This program is definitely *not* workaday: it is the first TDM plug-in to give musicians the ability to create custom processors in the form of new TDM plug-ins. In this, it is similar in principle to Cycling '74's *MSP* (reviewed in the October 1998 **EM**), which allows the creation of VST plug-ins, albeit not to the same degree.

Working on a black surface (shockingly called the Blackboard), the user can connect any of 40 modules using virtual patch cords to create complex patches, or presets, in a manner similar to patching a modular synth. Patches can operate as mono/mono, mono/stereo, or stereo/stereo, determined by the track into which they are inserted. A Patch Manager feature allows you to switch between patches automatically, which adds flexibility and helps you manage your TDM DSP resources.

DUY DIY

The modules (listed in the table "DSPider's Top 40") range in function from low-level math objects (add, subtract, multiply, and divide) to envelope followers, delays, spectral shapers, and simple reverb building blocks. You even get preprogrammed user-interface modules, such as sliders, plasma meters, numeric readouts, scopes, and scales. The modules reside in a palette to the left of the Blackboard, which also houses the Patch Cord and Eraser tools (for connecting and deleting modules, respectively) as well as Save and Load buttons for storage and recall of patches to and from disk.

Pop-up windows containing programming parameters can be opened by double-clicking on some of the more complex modules. Although this approach makes sense, these pop-up windows are modal, meaning that the program will not let you to do anything else while they're open, which is a bit inconvenient.

DSPider has two modes: Reader and Advanced. In Reader mode, preexisting *DSPider* patches can be recalled and used, but not edited; the "guts" of the patch—the modules and their connections—remain hidden. Advanced mode gives you a larger work area, contains the module palette, and allows patch creation and full editing, except with patches that have been "locked."

Locking a patch hides its component modules so that other users cannot see its design. This was intended to aid those wishing to develop *DSPider* patches for commercial sale; Opcode's *Max* (a MIDI/DSP application-development environment that uses a similar module/patch-cord design concept) has a comparable feature.

Much to my surprise, *DSPider* lacks a pitch-change module. However, according to DUY, such a module is in the works.

INTERFACE MODULES

Because *DSPider* is a tool for creating software, its user-interface modules (such as Sliders, Meters, Scopes, and Scales) are essential, and they have been graphically designed to be space-efficient and visually pleasing. There is one rather glaring omission: the modules include no knobs. The only module that emulates a physical control is a resizable slider.

Needless to say, the absence of knob controls severely limits the interface design possibilities for patches, in that functions that would best be controlled by knobs must instead be controlled by sliders. Something like Emagic *Logic Audio*'s Auto knobs would be helpful here; these controls change between a knob and a slider, depending on how they are resized. Having knobs would also be an advantage in allowing for more efficient use of screen real estate, since *DSPider*'s Blackboard can extend only to a certain size.

Of course, *DSPider*'s Sliders are certainly useful, and one does become accustomed to them. But I particularly liked the Scopes, which produce a scrolling, 3-D, continuous oscilloscope-like display when analyzing an input source. They are immediately useful when you want to troubleshoot or use a *DSPider* patch.

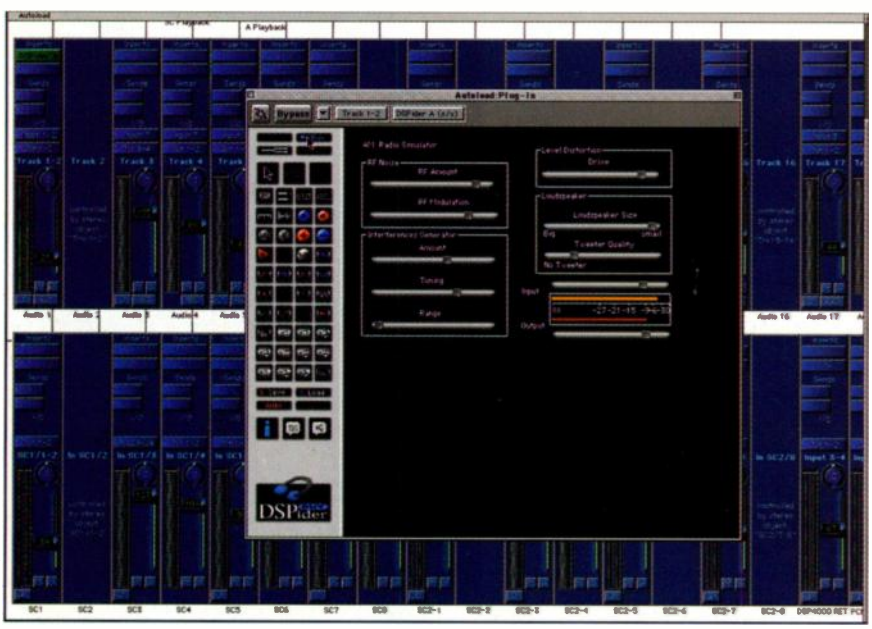

FIG. 1: DUY's *DSPider* comes with 40 different modules and 170 patches. This AM Radio patch makes the source material sound as if it were being played by a single-speaker AM radio. The source sounds smaller and is partially obscured by realistic hash and distortion.

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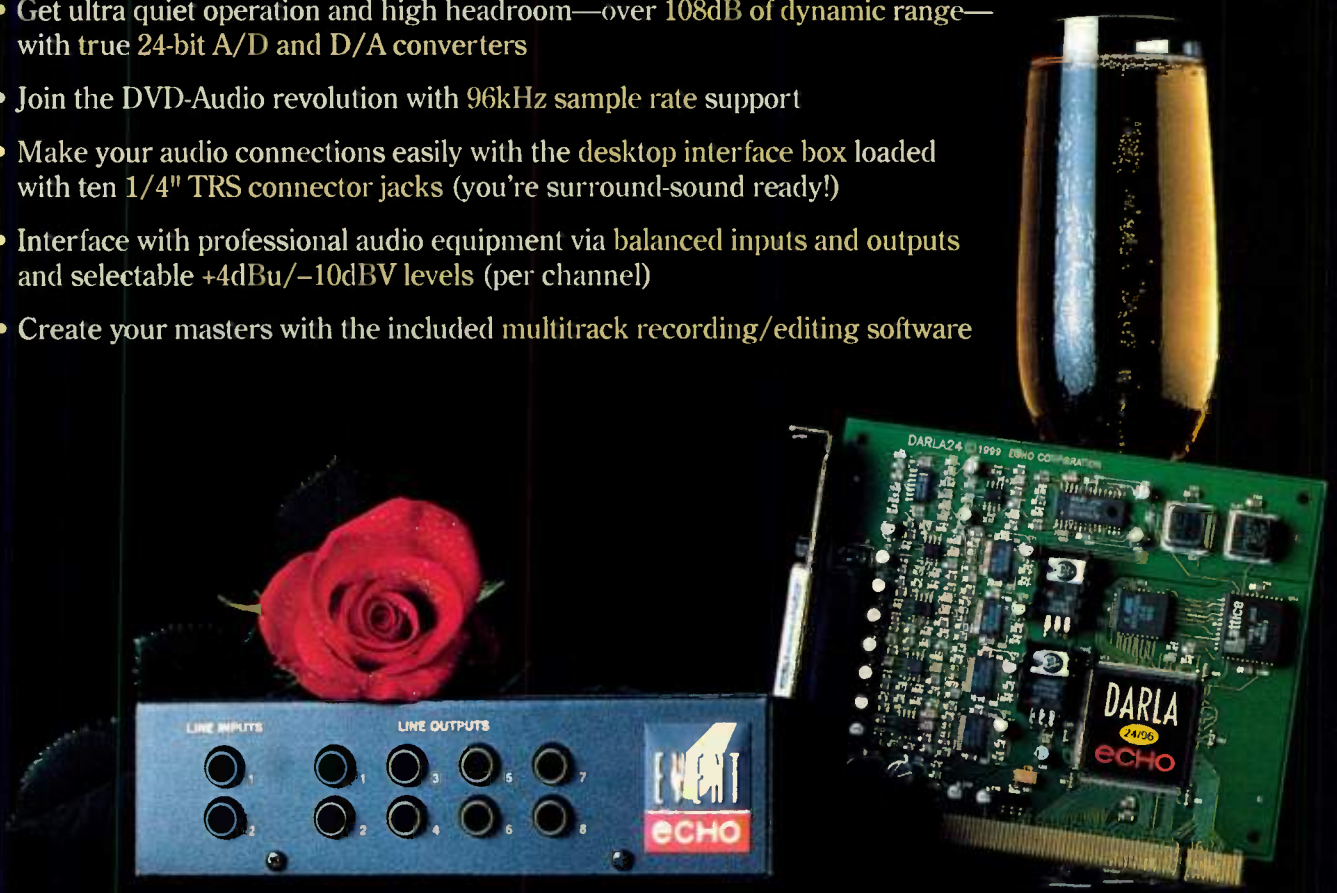

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DANGER, WILL ROBINSON!

The math modules (such as Invert, Addition, and Subtraction) also are essential. Because of the nature of digital audio, performing simple mathematical operations (sign changes, addition, multiplication, and so on) is frequently necessary to achieve a desired result. The test patch I'll describe shortly is a case in point.

The math modules have to be used with caution because of the potential for speaker and hearing damage. *DSPider* provides low-level control over digital audio signals, with no safeguards of any kind, and the innocuous-looking math modules can produce extreme and unpredictable results if you don't know exactly what you're doing. Disaster can strike easily and in a number of ways; for example, dividing by zero produces the maximum output possible.

Although warnings to this effect are provided at various places on the package, they should be prominent and more frequent in the manual. I'd like to see DUY come up with something unmistakable, like a large, red skull-and-crossbones icon, or perhaps a small mushroom-cloud or biohazard symbol to prevent people from damaging themselves and their equipment. A pop-up alert would help: "Warning! You are about to divide by zero. Are you sure you want to do this? Turn down your monitors!" It's unrealistic to assume that most musicians using the program have a background in digital audio theory—or read manuals carefully.

FAT-FREE FILTERS

I was somewhat disappointed in *DSPider*'s filters, in terms of both functionality and sound quality. I was hoping to find at least one solid, flexible, and sonically powerful multimode-filter module. Instead, I was presented with a few single-mode, rather digital-sounding offerings.

I would have preferred just one or two multimode-filter types, but *DSPider* has six separate filter modules. While I realize that this was probably done in the interest of DSP-resource efficiency, using radio buttons to select from six filter modes in a single module would have been more elegant for the user.

The sonic issues are much more important, though. Although digital filters aren't known for their fat, rich sound, good-sounding digital filters can be found on many hardware and software products, ranging from Clavia's Nord Lead synths to E-mu samplers and Eventide's DSP4000 effects processor, just to name a few.

In contrast, to my ears, the filtering in *DSPider* is clinical and thin, lacking the richness and musicality needed to produce inspiring results. This is particularly ironic because DUY has made its name with software that successfully emulates some of the characteristics of analog systems, such as tubes and open-reel tape. In light of this, I expected more from these filters.

LESSONS LEARNED

You'll find the most success with *DSPider* if you begin with a clear idea of what you want to create. In my case, the first thing I sat down to make was a simple stereo panning filter with a single LFO controlling the pan. I figured that this would make an effective test project that would provide insight into how easy (or difficult) it is to build things from scratch, without being horribly complex.

My filter design actually called for two separate mono filters that would sweep in contrast to each other, so I dragged the following onto the Blackboard: two 2-pole lowpass filters, one oscillator (to serve as an LFO, using the LFO setting on the control sliders), an inverter (to invert the LFO's output for inverse sweeps), some sliders for control, and meters for output level. I also dragged out some Mixer modules so that I could combine the manual filter-slider control outputs with the LFO outs for each side of the filter.

After connecting these elements, I was dismayed to find that instead of the nice stereo filter sweeps I was after, I was rewarded with loud, piercing, high-frequency sweeping tones instead. My source signal (a CD in this case) was buried in the background.

After some communication with DUY, I realized my mistakes. First, I had neglected to change the LFO outputs to positive values (negative values make no sense in this digital-domain context). Second, I should have placed multipliers at the outputs of the mixers when combining the control signals from the sliders and LFO.

Although the manual mentions the issue with negative numbers, I'd managed to miss it (proving my earlier point about musicians and manuals). But I would never have thought to use the multipliers postmixer. This is a prime example of a situation for which tutorials would have been invaluable. Sonically, the end result of my filter project was less than thrilling, but it did function as planned—*after* I'd sought help from DUY.

DSPIDER'S TOP 40

DSPider comes with 40 modules for programming custom patches. More modules are being developed, including a pitch shifter and a new reverb, but as of this writing, the program includes:

- Absolute Value
- Addition
- Early Reflections Chamber
- Envelope Follower
- Invert
- Long Delay Lowpass
- Long Sample-Modulated Buffer
- Long-Delay All Pass
- Medium Delay All-Pass
- Medium Delay Lowpass
- Medium Sample Buffer
- Mixer
- Multiplication
- Noise Generator
- Numeric Readout
- One-Pole Lowpass and Highpass Filters
- One-Sample Delay
- Oscillator
- Pitch Tracker
- Plasma Meter
- Ramp Generator
- Sample and Hold
- Scale
- Scope
- Shaper
- Shift Left
- Shift Right
- Short Delay All-Pass
- Short Delay Lowpass
- Short Sample Buffer
- Slider
- Spectral Shaper
- Subtraction
- Text Label
- Triangle Oscillator
- Two-Pole Band-Reject and Band-pass Filters
- Two-Pole Lowpass and Highpass Filters

PLENTY OF PATCHES

Version 1.0 of the program included two disks of patches (approximately 170 patches in all). The version 1.1 upgrade adds 50 more patches, which have been optimized for the superior DSP power of the Pro Tools 24/Mix and Mix+ systems. The patches are organized into folders by category: Chorus/Flangers, Compressor-Expanders, Delays, Distortion, Equalizers, Exciters, Experimental, Limiters/Levellers, Mastering, Miscellaneous, Multi-Processors, Noise-Reduction/Gates, Reverbs (long and short), Sound Effects, Spatial/Pan/3D, and Synthesis. A complete list of patches can be found on DUY's Web site.

Some of these patches are very interesting. AM Radio, in the Experimental category, is fun and potentially quite useful for sound design (see **Fig. 1**). This AM-radio simulator has controls for RF Noise, Level Distortion, and Loudspeaker Size, and it includes an Interference Generator that has Amount, Tuning, and Range parameters. It does just what you'd think: your source is partially obscured by a haze of very realistic hash and distortion, and the sound is generally made "smaller."

One of my favorites is Re-Tape (see **Fig. 2**), which simulates a faulty tape echo, complete with adjustable "tape jumps" (momentary speed fluctuations) and delays of up to about 185 ms. I loved the atmospheres it created, especially on guitar and percussion.

A patch that seems to be the sonic cousin of DUY's *Shape* equalizer is the 3-Band Smooth Stereo Equalizer, which had the same radical range of cut and boost that *Shape* does, but with a vastly simpler interface (see **Fig. 3**). Though it's not ideal for razor-precise notches or boosts, I like it very much for more general, overall shaping. It is indeed smooth, as its name implies, and not too clinical.

Back on the less conventional side of things, I enjoyed working with the Stereo Ring patch quite a bit. A simple dual ring modulator, it's capable of some gleefully apocalyptic sound destruction, and it's well worth playing with.

On the subject of sound destruction, the Distortion category holds some cool creative tools, notably the Distortion Synth (envelope-controlled distortion

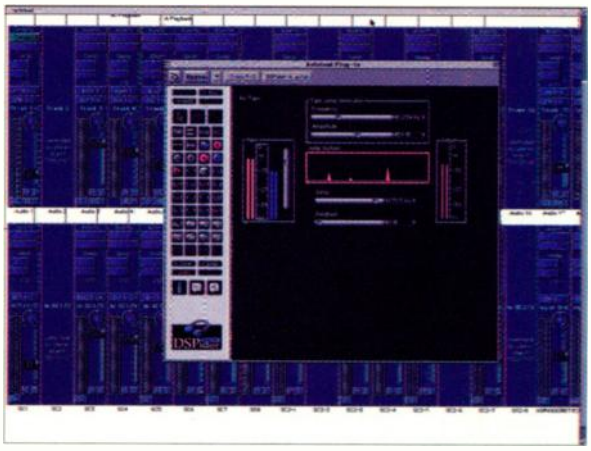

FIG. 2: The Re-Tape patch simulates a faulty tape echo, complete with adjustable momentary speed fluctuations. It's especially nice on guitar and percussion.

with filtering) and Mains Distortion (somewhat like Digidesign's *Recti-Fi*). The amounts of distortion and noise in Distortion Synth vary depending on the input level. Mains Distortion can generate hum and noise that can be blended with the distorted/rectified input signal.

I also liked the Dual Noise Gate patch, mostly because of the smoothness of its response. This was a surprise—somehow, I didn't expect *DSPider* to excel in this area. But I got good results on sources ranging from drums to vocals.

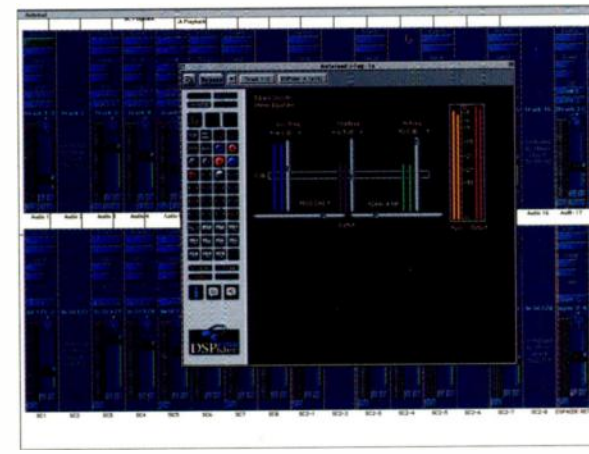

FIG. 3: The 3-Band Smooth Stereo Equalizer allows extreme cutting and boosting and is particularly useful for general sound shaping rather than high-precision tweaks.

I wasn't impressed with the reverb patches; they tend to sound cold and "electronic." This shortcoming needs to be addressed, because reverb is one of the most commonly used effects, and customers will expect a good one. Fortunately, DUY acknowledges the problem and has assured me that this is being worked on. Hopefully, they'll get it right the second time.

DOCUMENTATION DEJA VU

From the beginning, reviewing *DSPider* gave me a feeling of déjà vu. Ten years ago, I reviewed the first release of Digidesign's *Turbosynth* software synthesizer for the Mac, which could be considered *DSPider*'s ancestor. Like *DSPider*, *Turbosynth* uses a module/patch-cord paradigm to allow either synthesis from scratch or sample manipulation onscreen. Its documentation suffered from the same main flaws *DSPider*'s does.

For one thing, the basic descriptions of the program's modules provided only cursory function information. They didn't tell you why the modules behaved the way they did, or (more importantly) how they could be used most effectively. Apparently, Digidesign had assumed that the user would have experience with conventional modular synthesizers as well as a technical or engineering background—which wasn't (and isn't) the case with the majority of musicians.

In addition, *Turbosynth*'s original manual included no applications chapter. Nor was there any extensive tutorial with example patches to illustrate the nuts and bolts of constructing patches and explain why they worked the way they did. These were major omissions, not only because the program represented something of an innovation and was unfamiliar to most potential users, but also because any system with this level of complexity requires thorough documentation to be used effectively. The *DSPider* package has both of these problems.

According to DUY, an in-depth book about *DSPider* programming is currently in the works and may be available around press time, but the company opted to release the program without it. If *DSPider* were a less complex product, this wouldn't be a big deal, but in this case it can easily leave users "lost in the woods."

WISH LIST

In addition to problems with documentation, the absence of a pitch shifter, and a lackluster reverb, *DSPider* has a number of other significant foibles.

Because of the small size of the modules and the complexity that is easily achieved within a patch, the process of patch creation can get confusing. Look

DSPider 1.1
Minimum System Requirements
Power Mac with Digidesign *Pro Tools* 4.2 or later with TDM (including Pro Tools 24/Mix); DAE 3.2.3; at least one free chip on DSP Farm card

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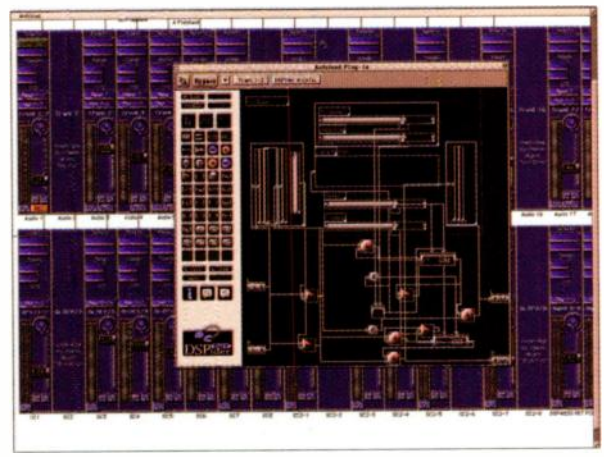

FIG. 4: A look inside Re-Tape reveals a virtual maze of connections and modules, and you can create far more complex *DSPider* patches than this one. The absence of color coding and other distinctive visual cues sometimes makes it difficult to see what's going on.

at the guts of the Re-Tape patch (see Fig. 4), and you'll see what I mean—and this is by no means an extraordinarily complex patch. To help alleviate this mess, I'd like to see DUY implement color coding to distinguish between, say, audio and control lines in a patch. Apparently, this was discussed at DUY during development, but it was nixed because in some situations the program would have no way of distinguishing between control lines and audio lines. One solution might be modifier-key patching; for example, holding down the Option key while patching could designate a connection as a control line, and using the Command key could designate an audio connection.

In addition, the icons for the modules, though tastefully designed, are a bit too subtle at times. This can be confusing to the eye when working with a complex patch. Perhaps a bit more visual distinction between the modules, as well as brighter colors, would rectify this.

I wish the Blackboard and the whole *DSPider* window were freely resizable to any size (including the full screen on a 20-inch monitor). I would also like to have zoom tools for zeroing in on sections of a patch. The screen clutter and confusion factors add up quite easily, especially with complex patches. It's hard to make sense of a densely packed window full of small modules and a labyrinthine forest of patch cords. This is another area in which DUY could learn something from Emagic: *Logic Audio*'s object-oriented Environment page used to become unmanageable frequently, but finally organizational tools (especially display filters) were added to make viewing specific object types possible.

There is no way to handle keyboard shortcuts in *DSPider* because the program runs under a host application, and some host applications do not allow key commands to be passed on to a plug-in. Other host applications, however, do support keyboard shortcuts for plug-ins, and I'd like to see that implemented here. To my mind, keyboard shortcuts are a necessity, particularly for tool selection (and an Undo function, if one were added).

ARE WE THERE YET?

I can't think of a better, recent "textbook" example of a potentially great and powerful product being sent to market before it was ready. *DSPider* seems unfinished and has the general air of a product in its infancy. A better manual, a more complete selection of

DUY
DSPider 1.1 DSP plug-in
$1,495

FEATURES ■■■
EASE OF USE ■■
SOUND QUALITY ■■▮
VALUE ■■
1 2 3 4 5

PROS: Vast potential for sound creation and processing. The only TDM plug-in of its kind. Multifunctional and endlessly upgradeable.

CONS: Current version lacks pitch processing and high-quality reverb. Interface can be difficult and confusing. No knob-controller module. Inadequate documentation. Overpriced, considering its current state of development.

CIRCLE #444 ON READER SERVICE CARD

modules, and comprehensive tutorials are all sorely needed if *DSPider* is to attract and keep a serious following. Furthermore, the software is expensive. For almost $1,500, I expect a fully developed and complete package.

As mentioned earlier, *Turbosynth* suffered from many of the same problems, but it eventually went on to gain a loyal cult following and is still in use today. DUY still has a lot of work to do, but given its fine track record, I strongly suspect the company will develop *DSPider* into a mature and elegant tool.

One late-breaking piece of news is that a read-only version of *DSPider*, called *ReDSPider*, will be available for $549 by the time you read this review. That might represent a more cost-effective option for those people who don't need *DSPider*'s patch-creation abilities. Both *DSPider* 1.1 and *ReDSPider* can run on Digidesign's new Pro Tools 24/Mix system.

Whether you should purchase *DSPider* in its current version depends largely on what you expect to gain from using it. If you're looking for a sort of Swiss army knife–style, multifunction TDM plug-in with a vast library of preexisting, world-class patches, you may have to wait awhile. It's entirely possible that DUY and third parties will eventually develop a patch library of this size and quality, and as noted, DUY is already working on a pitch shifter and better reverb. But we're not there yet. Even when the library expands, the user who does not need to create original plug-ins should focus on *ReDSPider*.

But if you are serious about *creating* patches and are willing to put in the time to learn *DSPider*'s intricacies, you can do a lot of interesting things right now. TDM users are likely to reap significant long-term rewards if DUY improves and extends the program's capabilities as promised. *DSPider* obviously has considerable sonic and creative potential, as evidenced by the included patches. In the long run, it could give your Pro Tools system amazing sound-mangling power.

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, Nile Rodgers, Shawn Colvin, L. Shankar, Sussan Deihim, Richard Horowitz, and Seal.*

EVENT ELECTRONICS

TRIA

Crystal clarity through the magic of triamped monitoring.

By J.J. Jenkins

First impressions can be a funny thing. Some people, myself included, would like to think that they're not easily swayed by every pretty piece of gear that comes their way. After all, it's what's inside that counts. Nevertheless, upon removing the Event Electronics Tria (triamplified) system from its ample shipping carton, I was immediately infatuated—it looks great, sporting a classic design and the look of raw power.

A quick overview of the system revealed that it has three major components: a subwoofer and two satellite speakers. The subwoofer box, or Very Low Frequency (VLF) station, has an 8-inch polypropylene cone. The VLF also houses the power amplifiers, three active crossovers, and the trim and level controls for the entire system. The two satellite speakers each come equipped with a 5¼-inch polypropylene cone driver and a 1-inch silk dome neodymium high-frequency driver.

Placement of the system components is extremely important. Although the satellite speakers effortlessly took their stations above my console, I had trouble finding a spot for the subwoofer box. Traditionally, and as the manual recommends, the VLF is meant to reside on the floor—ideally somewhere horizontally between the other two speakers. Unfortunately, because of the way my studio is designed, there is no unobstructed spot under the console. I tried the VLF out in a few different places in the room before settling on the shelf above the console between the satellite speakers. Lifting the unit up there turned out to be less of a chore than I had anticipated: weighing in at 34 pounds, the VLF is lighter than other speakers I've hauled up there. The satellites themselves are compact and light, each weighing only 11 pounds.

The satellites have tiny, removable feet that act as tilt stands and are designed to facilitate on-axis positioning. The small size of the satellite speakers makes the Tria a good candidate for multimedia work: each of the components is magnetically shielded, so you can use the system with computer workstations and video monitors.

The layout on the back of the VLF component is straightforward. The audio inputs utilize Neutrik combo connectors, which can accommodate either a balanced XLR or an unbalanced or balanced ¼-inch feed. Each satellite speaker is connected to the VLF with a multipin cable, which looks very classy.

The VLF is equipped with compact and recessed potentiometers for adjusting input sensitivity. There are also trim controls for the satellite speakers' high and low drivers, and one for the VLF's subwoofer. The pots are easily turned with a screwdriver or a long fingernail. The only complaint I have with the Tria system is that these controls are on the back of the VLF, which makes tweaking the system a bit of a hassle. You can, of course, remedy the situation by positioning the VLF with its back facing toward you.

EVENT ELECTRONICS
Tria triamplified monitoring system
$999

AUDIO QUALITY ■■■■■
VALUE ■■■■■
1 2 3 4 5

PROS: Great clarity. Reasonably priced. Satellite speakers are small. Adjustable trim for three frequency ranges.
CONS: Rear-side controls on VLF are difficult to access.
CIRCLE #445 ON READER SERVICE CARD

The Tria triamplified monitoring system from Event Electronics consists of the VLF and two satellite speakers. The system fits easily into almost any room and reproduces material with amazing clarity.

PLAYING AROUND

I started making initial adjustments by tweaking the left and right input-sensitivity levels. The input sensitivity has a 20 dB range, providing a fair amount of room to maneuver. (You'll want to calibrate the system with the master level on your console.) The satellites are equipped with green LEDs that glow constantly when the speakers are powered up and flash just before clipping occurs in the amplifier.

The tonal trim controls give you a great deal of flexibility in tailoring the system to suit your room. The high frequencies are separated at 2.9 kHz with an active second-order crossover, and the ultra-lows are separated at 60 Hz with a third-order crossover. All three frequency ranges offer ±3 dB of gain. After trying various combinations, I ended up with all the settings pretty close to 0 dB, achieving a relatively flat sound.

While I moved the VLF to several different positions in the room before settling on a location, I was amazed at the consistency in sound. I seldom had to

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Tascam tel. (323) 726-0303; fax (323) 727-7635; Web www.tascam.com

Tria Specifications

Frequency Response	35 Hz–20 kHz (±3 dB)
Power Rating	VLF: 80W continuous; LF (satellites): 80W continuous per side; HF (satellites): 40W continuous per side
Input Connectors	(2) Neutrik combo (XLR or ¼")
Input Sensitivity Adjustment	20 dB
High-Frequency Driver	1" soft dome
Low-Frequency Driver	5.25" polypropylene cone
Very-Low-Frequency Driver	8" polypropylene cone
Crossover Frequencies	2.9 kHz, active second-order (satellites); 60 Hz, active third-order (VLF)
Low-Frequency Trim	±3 dB @ 100 Hz, ±2 dB @ 400 Hz
High-Frequency Trim	±3 dB above 2.6 kHz
Dimensions	Satellites: 7.5" (W) x 10.5" (H) x 9" (D); VLF: 12" (W) x 18.5" (H) x 11" (D)
Weight	Satellites: 11 lbs. each; VLF: 34 lbs.

adjust the low-frequency trim control to balance out the response. The only time I had to roll off the low end was when I put the VLF on the floor in a somewhat unusual location. I would have tried pushing the envelope further in terms of positioning, but the satellite cables are only 10 feet long.

Tria's power amplifiers definitely pack a punch. The system utilizes the same technology as Event's 20/20bas monitors to power the satellites, with the highs driven at 40W and the lows at 80W. The VLF is powered by a separate 80W amplifier. The amplifiers all offer plenty of headroom and are very clean. There is a circuit breaker located next to the on/off switch on the back of the VLF unit in case of unforeseen circumstances.

THE PAYOFF

I listened to several pieces of music that I am intimately familiar with, specifically looking for some sort of coloration, and I found the speakers to be totally transparent. I was especially impressed with the system's clarity and separation. Overall, Tria sounds great.

And let's not forget the price. I had no idea what the list price was until late into this writing, and I was amazed to find that it's $999. I had thought it would be a bit higher, so I was pleasantly surprised.

First impressions—I wouldn't bank on them every time, but sometimes they can be right on. I can't wait to hear a 5.1 version of this system at some point in the future.

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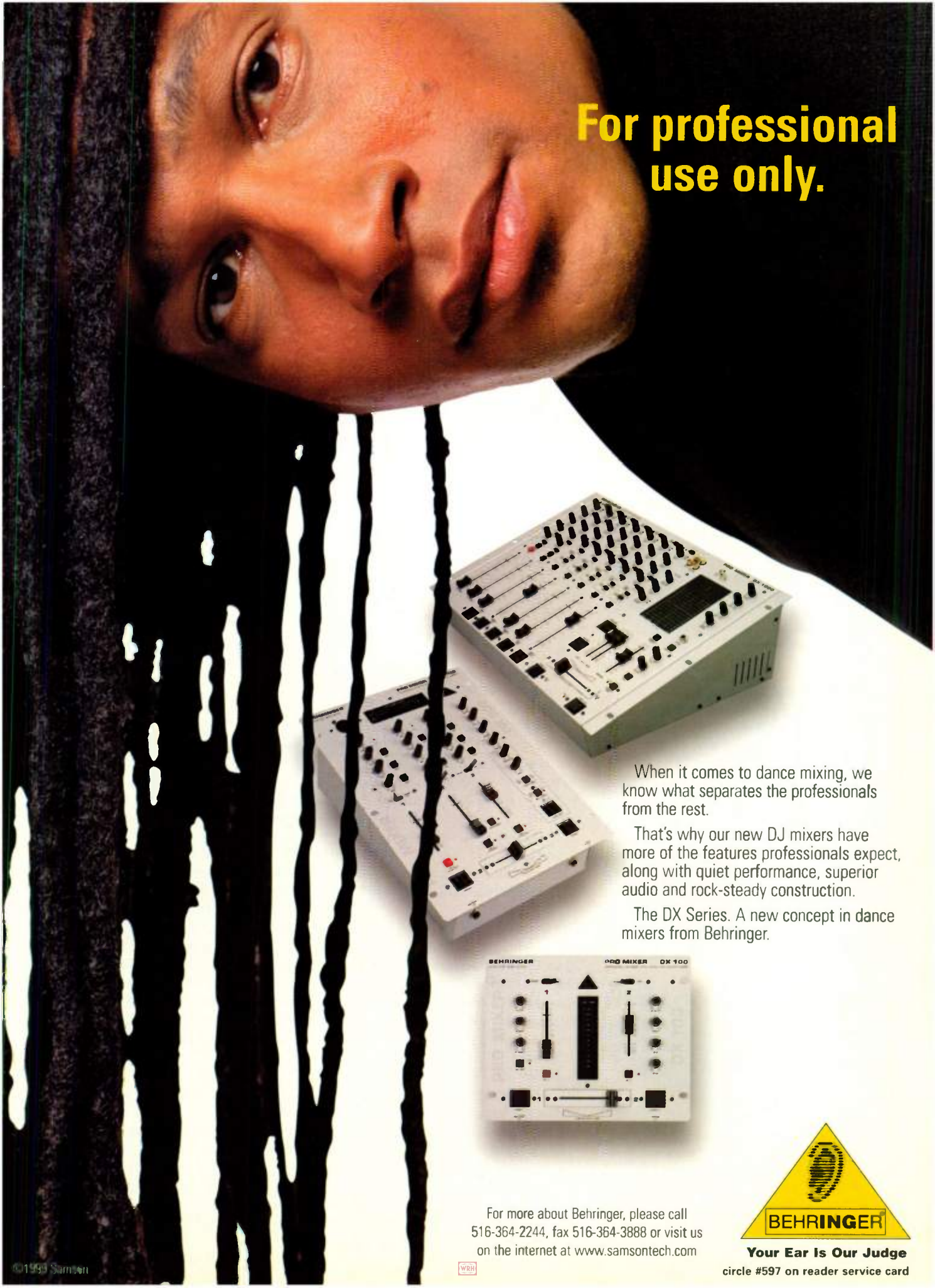

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QUICK PICKS

TASCAM
DA-45HR

By Gino Robair

As a person who frequently records direct to DAT, I've been eagerly awaiting the chance to upgrade from 16-bit to 24-bit resolution. Enter Tascam's 24-bit DAT machine, the DA-45HR ($2,165).

High Speed

In order to get 24-bit word length onto a DAT tape, the DA-45HR records at twice the normal speed of a 16-bit machine. That means you get 60 minutes of recording time from a 120-minute tape. Tascam recommends against using 180-minute tapes because of the thinness of the tape stock, so maximum recording time at 24 bits is one hour.

The DA-45HR can also function as a conventional 16-bit unit. You can record in either Standard mode (16 bit) or HR mode (24 bit, double speed). During playback, the machine automatically recognizes the tape's bit rate, so having to switch is unnecessary. The DA-45HR supports both 44.1 and 48 kHz sampling rates in either bit resolution.

The Inside Track

The DA-45HR has 24-bit A/D converters and 20-bit D/A converters, and three dithering options are available (triangular, rectangular, and off). Analog connections are on balanced XLR and unbalanced RCA jacks, and the digital connections are AES/EBU (XLR) and S/PDIF (RCA coaxial). Other features include a BNC jack for word clock I/O, a data/shuttle wheel, and character editing and titling. An optional remote control, the RC-D45 ($99), is also available.

One drawback of the DA-45HR is that it has a noisy fan. Presumably, the fan keeps the interior cool when the unit is in HR mode. The volume level of the fan is comparable to that of a computer or disc drive: just loud enough to be noticeable. Tascam is working to correct the problem with a quieter, shock-mounted fan in future versions of the unit. In the meantime, however, they have made available a free do-it-yourself modification kit for current DA-45HR owners.

In the Field

The DA-45HR operates like any other DAT machine, so getting up and running is easy. First, I compared 16-bit recordings made with the DA-45HR and a Tascam DA-30. There was a noticeable difference between the two: the DA-45HR produced a smoother, rounder sound with a fuller low end than the DA-30. Using the DA-45HR in HR mode (24-bit recording/20-bit playback) yielded even more detail compared with its own 16-bit recordings. The previous takes sounded harsh in comparison.

To get full 24-bit performance (A/D *and* D/A) from the DA-45HR, I used an Apogee PSX-100 digital converter. Tracks made with the Apogee's 24-bit ADC and played back through the DA-45HR's 20-bit DAC showed an increase in clarity over the material recorded with the DA-45HR's own ADCs. However, using the PSX-100 at both ends made the biggest difference. I was rewarded with an improved dynamic range, quicker transients, greater low-level detail, and a smooth, even blend throughout the frequency spectrum.

The Tascam DA-45HR is a 24-bit DAT machine with 16-bit backward compatibility.

Sweet Sounds

Overall, the DA-45HR is a definite improvement in DAT technology. In Standard (16 bit) mode, the recording quality is noticeably better than that of the average DAT machine. Because of its backward compatibility and price, the DA-45HR is a good choice if you're thinking about adding a DAT deck to your rig.

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The real payoff, however, is in the detail you gain by using the DA-45HR in HR mode. Even though the unit's internal D/A converter is only 20 bit, it still delivers a noticeable increase in dynamic response and overall resolution over 16-bit recordings. Add an external 24-bit DAC and you can take full advantage of the DA-45HR's potential. If your studio and DAW are moving in the direction of 24 bits, or if you want to make the first step into the world of higher resolution, the DA-45HR is an excellent choice.

Overall EM Rating (1 through 5): 4.5

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BEST SERVICE

XX-Large Pads

By Jeff Obee

A two-CD sample set can contain up to 1,300 MB of audio files, and *XX-Large Pads* ($99.95, double audio CD; $199.95 Akai, Emu, and SampleCell format CD-ROM) from Best Service uses practically every available byte in this offering. If you enjoy a lush pad as much as I do, you'll be spending considerable time in front of your sampler with this collection.

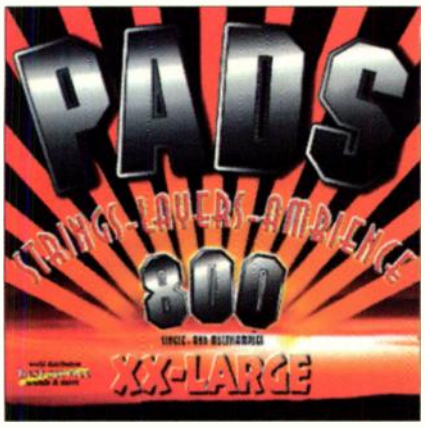
Large Sounds

XX-Large Pads features more than 800 samples, which are divided into three categories: Single Chord, Digital MultiSamples, and Analog Pads. There are lush, layered pads (some of which have chorus or phasing applied to them), strings, and ambient washes; all are stereo samples, and the Analog section has some mono samples, as well.

The Single Chord sounds are all sampled on middle C, so there is a limited range available. Because these sounds are generally used as washes or drones, however, it's not a problem. The latter two categories are sampled either every octave or in fifths, from C1 to C6. A short (and not necessarily apropos) demonstration of the sound is given before each sample in these categories, so you know what you're getting into.

Walking on Air

The synths sampled on the discs include the Kurzweil K2500, Korg Trident and Wavestation, Oberheim OB-8, Waldorf Wave, Prophet VS, Roland JV-1080, and others. There is an emphasis on airy pads and string sounds. The type of atmospher-



***XX-Large Pads* from Best Service features an abundance of useful synth flourishes and background textures.**

ic wavetable patches that we have heard so much throughout the '90s (typically a central tone augmented by ambient flourishes) are here in abundance.

Among the Analog samples are some filter-swept pads and thick, sawtooth-wave patches that stand well on their own or provide an excellent foundation for tweaking, especially if you own a sampler with an onboard synth engine. One of my favorite textures is Fantasy Gong, which I processed through a lowpass filter and drenched in a chorus/reverb effect for what became a beautiful background texture.

All of the samples were recorded using a Roland S-760. They are uniformly clean and excellent sounding: there are no clicks, extraneous noises, or variances in quality throughout the set. Some of the samples have abrupt cutoffs, however, so you may need to do some looping. That can sometimes be a difficult task with sounds like these if you're using the audio CD; if you have the cash, the CD-ROM might be a better investment.

Judgment Day

I liked this set quite a bit. It's reasonably priced, and there's a versatile selection of programs. The documentation is negligible, however, and samples on each track aren't referenced according to exact time and length. Also, the Single Chord patches have no names, which would be helpful considering there are from four to seven sounds in any given track. All of these issues are minor, though. The fact is, if you get a quiver down your spine when a deep, alluring wash caresses your ears, you should give *XX-Large Pads* a listen.

Overall EM Rating (1 through 5): 4

CIRCLE #447 ON READER SERVICE CARD

QUICK PICKS

EAST WEST/MZONE

Groovemasters Drums

By Jeff Obee

Mads Michelsen may not be familiar to most of us here in the States, but he is a well-known cat in Denmark and northern Europe. His tight, crisp drumming is the focal point of *Groovemasters Drums*, a double CD set released jointly by East West Communications and the Danish company Mzone. I reviewed this collection of samples on audio CDs ($99.95); it is also available on CD-ROMs ($199.95) formatted for Akai and NemeSys *GigaSampler*.

Method to His Mads-ness

Groovemasters Drums is neatly divided into loops and fills (one or two bars each) on disc 1 and single drum hits on disc 2. For the sake of definition, the styles are delineated as Rock, Rock & Roll, Shuffle, Straight, Disco, and Mixed. You also get R&B stylings, brush patterns, classic-rock grooves, roots/reggae rhythms, and much more. The Disco tracks are not restricted to the "pea soup" hi-hat beats you would expect (although that specific disco style is included); rather, "disco" is a blanket term that covers a multitude of dance grooves.

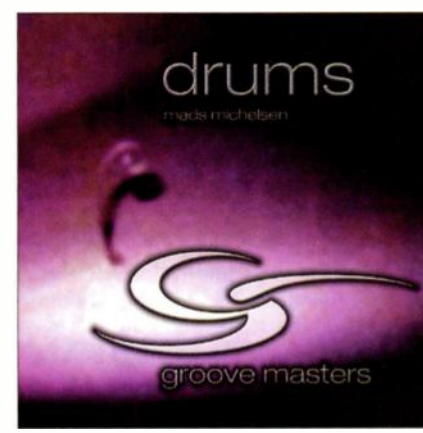
The material on the first CD is presented in fixed and free tempi. In the fixed section, Michelsen plays to a click, and in the free-tempo loops, he grooves without one. Each basic loop is followed by several alternative takes. For instance, a steady, simple rock beat may be followed by a version with a "pressed" or dragged snare, side stick, hi-hat variations, tom fills, ride cymbal, and so on. Plus, a subsequent version without bass drum is almost always included, too. I applaud the plenitude of variety and nuance on these tracks.

The same sentiments apply to the single-hits disc. You get an ample selection of bass drum, snare, hi-hat, cymbals, and tom hits at hard to soft velocities, with some nice variations like a tom with sympathetic vibration from the snare or played with a bend of the drumhead, a selection of China cymbals, and a humorous "Danish-Finish." The snares are sampled with ruffs and drags for more diversity and realism. A mallet jazz kit with tom rolls is also available in case you're ever in the mood for sequencing some Gene Krupa. My only complaint is that the hits go by a bit too quickly in some places.

Each track is indexed in detail, listing style, time location, the kit used, and so forth. The documentation is good and gives the user all the facts needed to make full use of the discs.

Key Components

Michelsen used three drum sets for these performances. The Rock set is a Drum



The first volume of the *Groovemasters* series, from East West and Mzone, is a collection of drum samples and loops with a well-rounded assortment of styles and sounds.

1999 Nominee

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Workshop kit, the Disco tracks use a Yamaha Professional Recording Series, and the Jazz kit is a Ludwig Vintage set. Some versions are recorded dry and others with ambient room mics.

On first listen to this CD I was struck by its tight, warm quality; upon examining the liner notes I discovered that it was recorded to analog tape, an excellent decision for recording a drum CD. The mics used include the Shure SM57; Neumann U 87, U 47, and U 69; Sennheiser 441; and AKG "The Tube" and D112; among others. These were sent through a slew of solid-state and tube compressors, and a Calrec mixing board, to an Otari MTR100 deck and Ampex 499 tape. A TC Electronics M5000 A/D converter was used to transfer to DAT.

A bunch of effects tracks were run through a Gibson Jazz guitar amp, and other tracks utilize an Eventide H3000 Ultra-Harmonizer, or a TC Electronics 1210 processor for delay and phasing effects. These unusual tracks really stand out. Some are dub oriented, and some are straight beats, while others are very syncopated and rhythmic (à la drum 'n' bass) for dance music. Disc 2 contains single takes of these effects hits for constructing your own off-kilter patterns.

The bottom line on any drum CD is the feel of the player, and Michelsen doesn't disappoint. He has a solid and simple yet imaginative approach; relaxed, yet aggressive and precise. His kits sound great—the single hits are outstanding—and his ideas are right on the money for such a production.

Mads About You

Being a bassist, I'd enjoy doing some gigs with Michelsen. That being unlikely, I will definitely use his loops and hits in my music. I had a delightful time sampling and toying with them. This is a great, "live" sounding set of CDs, the first in a series of specific instrument collections under the Groovemasters moniker that are intended to work together. I recommend that you check this one out.

Overall EM Rating (1 through 5): 4 .5

CIRCLE #448 ON READER SERVICE CARD

CAKEWALK

Audio FX 3 (Win)

By Mike Lawson

Cakewalk's *Audio FX 3* ($249.95) is a DirectX plug-in you can use to design custom ambiences for your digital audio tracks by building virtual tracking rooms from the ground up. It's easy to create a new space: simply click and drag your mouse to move walls and change ceiling height. Then position up to two "performers" anywhere in the room, and use the onscreen Trackball to adjust the listener's orientation to the space. The room sound is captured with virtual microphones, and you can adjust the microphones' positions and polar patterns, as well.

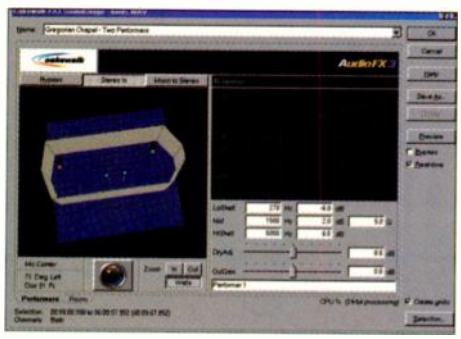

FIG. 1: In Cakewalk *Audio FX 3*'s Performers View, you position the performers and microphones in your custom-made virtual room.

In My Room

Audio FX 3's attractive interface toggles between two windows, the Performers View and the Room View (see **Fig. 1**). The Performers View displays your performers' positions in the room and provides settings to tailor the EQ curves for each performer. You can adjust the curves by manipulating settings on a graphic display or by moving sliders associated with each parameter (low-, mid-, and high-frequency EQ, for example).

In the Room View, you design the room your audio tracks will be performed in by defining the sizes and positions for ceiling, floor, and up to seven walls independently. This allows you to create rooms of any shape, from pentagon to hexagon to just plain square. Rooms can range in size from 20 to 200 square feet, and you can raise the roof on your space to a towering 200 feet. Other settings let you customize the room's absorption, trapping, and high-frequency damping characteristics.

The Room view also gives you access to five mic types, including omni, cardioid, and bidirectional. You can use two mics and separate them by as much as 100 feet. You can also adjust each mic's angle relative to the performer, from 0 to 180 degrees. A setting called Proximity Adjust Distance is also on hand to control the impact that moving a performer away from the mics will have on the reverberant quality of the signal. Normally, as you move the source away from the mic, the room sound becomes more prominent. But with *Audio FX 3*, you can counteract the laws of physics by using this setting.

Performance Issues

The system requirements call for a Pentium 166 with 32 MB RAM (a Pentium 200 is recommended for real-time stereo performance); however, I experienced some breakup when I used the plug-in in real time on a 200 MHz AMD-K6 with 128 MB RAM. (Adjusting the buffer settings in your host application could improve performance on a slower system.) Of course, once the effect was processed, I encountered no problems at all, and when I switched to an Intel Pentium II/350 computer with 128 MB RAM, *Audio FX 3* worked flawlessly throughout.

I really like this new Cakewalk release; it sounds great, it's easy to use, and it will expand your audio-processing options. If I could have only one room simulator, I'd want it to be this one.

Overall EM Rating (1 through 5): 4

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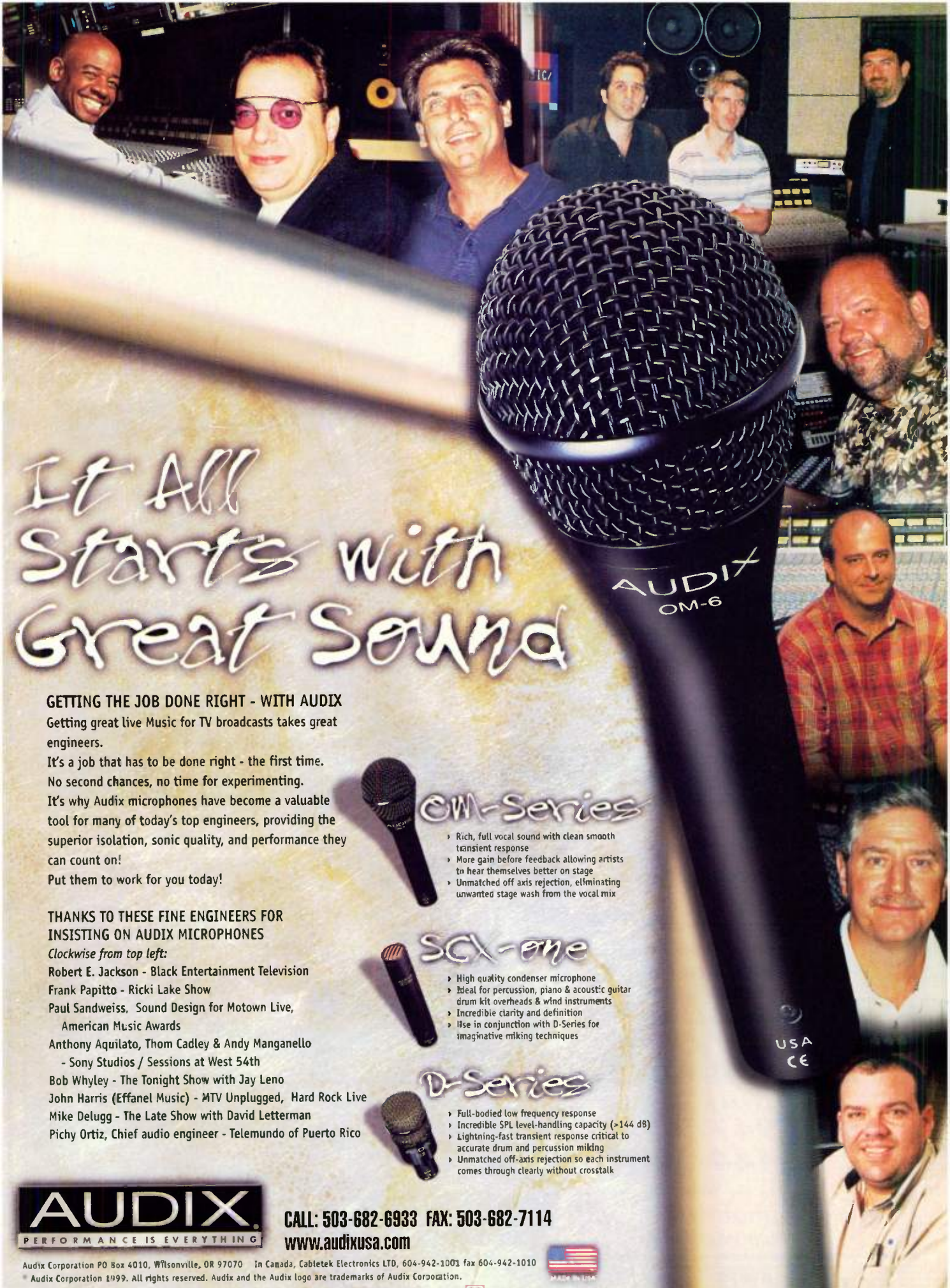

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The 1212I/O card helps bring the price of full function multi-channel computer based recording to a point that just about anyone can afford. It features 12 inputs and outputs configured as 2 analog I/Os, a S/PDIF I/O and and 8-channel ADAT optical I/O. All I/Os can be used simultaneously for maximum flexibility. Compatibility with most Digital Audio Software on the market and outstanding sonic quality make the 1212I/O a great choice for project studios and multimedia pros.

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Recording System

The Lexicon Studio System interfaces with your favorite digital audio software for a complete hard disk recording package. Supporting both PC and Mac, Lexicon Studio can be expanded up to 32 voices from a variety of I/O options. For recording, editing, mixing and DSP, Lexicon Studio is here.

FEATURES–
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digidesign Project II Bundle

A Division of Avid Technology, Inc.

The new Project II Bundle incorporates Digidesign's legendary SampleCell II card w/32MB RAM, MasterList CD and Logic Audio AV software together with the popular Project II recording card for a price that you are not going to believe! Just add your choice of interface and you have a complete PCI studio at your fingertips.

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882/20 20-bit interface

The new Digidesign 882I20 I/O is a high performance entry-level audio interface for Pro Tools. It features 20-bit D/A & A/D converters, 24-bit digital performance, and an extremely low noise floor. The 882I20 makes an excellent auxiliary audio interface for Pro Tools24 — ideal for connecting outboard signal processing gear, keyboards, or other external devices. It can even operate as a standalone, 2-channel, 20-bit A/D converter or D/A converter without Pro Tools (or any other) software.

ToolBox PCI Digital Audio Bundle For Mac or PC

When you need professional features at an affordable price, the Digidesign ToolBox delivers a great combination of software and hardware for Mac or PC. Based around Digidesign's AudioMedia III a 16-bit audio card with stereo RCA inputs, 1 bit 128x over sampling A/D and 18-bit D/A converters as well as coaxial S/PDIF digital I/O. This system is ideal for personal and project studios, radio broadcast applications, and multimedia audio production.

ToolBox For Mac Includes:–
- Includes **Audiomedia III** card
- **Pro Tools 4.x** recording/editing software with playback support for up to 8 tracks of audio
- **D-fx** AudioSuite Plug-Ins (Reverb, Delay & Modulation Effects)
- **D-fi** AudioSuite sound degeneration Plug-Ins
- **Bias Peak Le** 2 track editing software

ToolBox For PC Includes:–
- Includes **Audiomedia III** card
- **Session** Software recording/editing w/ support for playback of 8 tracks of audio • **Logic Audio AV** MIDI sequencing/ audio software
- **Sound Forge XP** 2 track editing software • **ACID Rock** loop based audio sequencer allowing you to dictate the pitch and tempo of any .wav file

SOFTWARE

Digital Performer 2.6
MIDI/AUDIO Software for Mac

MOTU

Their second major update this year, with a relentless stream of new advanced features, like sample-accurate editing, sample-accurate sync and MOTU's innovative RAM-based loop recording tool called POLAR. DP is packed full of features you won't find anywhere else.

FEATURES–
- Includes over 50 real-time MIDI and audio effects plug-ins • POLAR window - Interactive audio loop recording the way it should be • 24-bit recording and editing • 32-bit native effects processing - incredible sounding EQ and other FX • 64-bit MasterWorks™ Limiter and Multiband Compressor plug-ins included • Advanced waveform editor • Sample-accurate - the most reliable editing and tightest sync you can get • OMF export - transfer your entire session, crossfades and all, into Pro Tools • Samplers window - drag & drop samples between your Mac and your Sampler • PureDSP™ stereo pitch-shifting and time-stretching • Unlimited audio tracks, real-time editing, full automation and remote control • QuickTime digital video support, and much more • Compatible with Pro Tools|24, the MOTU 2408 and today's other popular systems • Digital Performer is an entire recording studio inside your computer

cakewalk Pro Audio 8
MIDI/Audio Software for PC

One of the industries leading MIDI/Audio software with a wide range of hardware support and realtime automation and editing for Windows 95/98 and NT 4.0.

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- SMPTE, MTC and MMC support
- Playback of .AVI, QT and MPEG video

ANTARES Auto Tune
Plug-In For Mac or PC

Intonation correcting multi-platform plug-in for Mac and PC considered to be the "Holy Grail of recording" by Recording magazine. Auto-Tune corrects pitch and intonation problems on voice and solo instruments without distortion or artifacts. Two modes of operation include Automatic where pitch is continously compared to a user selected scale and Graphical mode offering more precise control allows you to draw specific target pitches. Compatible with TDM, VST MAS and standalone on the Mac and DirectX or DAL V8 on the PC.

WAVES Native Power Pack

Uses the CPU of your Mac or PC to provide top quality effects processing for recording, mixing and multi-media applications. Compatible with many popular audio editing software programs, the NPP provides EQ, Reverb, Compression, Gating, Stereo imaging and the incredible L1 Ultramaximizer mastering peak limiter. It also includes Wave Convert, a stand alone application that batch converts formats, bit-depths & sample rates for the loudest, cleanest multi-media files available. A must have for recording engineers & internet designers alike.

Native Power Pack II

The all new Native Power Pack II is an entirely different plug-in collection than the original Native Power Pack. Bass-enhancement, de-essing, vintage compression/expansion and EQ are all provided, and can be used with or without the original NPP. You can also upgrade from either NPP or NPPII to the Native Gold bundle for the complete Waves experience, and like the earlier NPP, the NPP II ***requires no extra DSP!***

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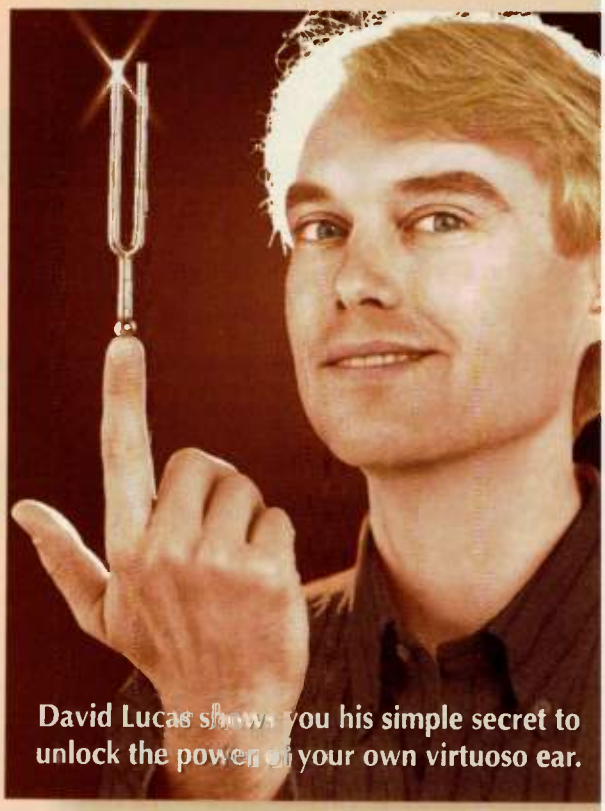

David Lucas shows you his simple secret to unlock the power of your own virtuoso ear.

by David Lucas Burge

IT ALL STARTED IN NINTH GRADE as a sort of teenage rivalry...

I would slave at the piano for five hours daily. Linda practiced far less. But somehow she always had an edge that made *her* the star performer of our school.

It was frustrating.

What does she have that I don't? I'd wonder.

Linda's best friend, Sheryl, sensed my growing competition. One day she bragged on and on about Linda, adding more fuel to my fire.

"You could *never* be as good as *Linda*," she taunted me. "*Linda's* got *Perfect Pitch*."

"What's Perfect Pitch?" I asked.

Sheryl gloated over a few of Linda's uncanny abilities: how she could name any tone or chord —*just by ear*; how she could sing any pitch she wanted—*from mere memory*; how she could play songs—after only *listening* to them on the radio!

My heart sank. *Her fantastic EAR is the key to her success* I thought. How could I ever hope to compete with her?

But later I doubted Sheryl's story. How could anyone possibly know F♯ or B♭ just by *listening*? An ear like that would give one a *mastery of the entire musical language!*

It bothered me. Did she *really* have Perfect Pitch? I finally got up the nerve, approached Linda, and asked her point-blank if it were true.

"Yes," she nodded to me aloofly.

But Perfect Pitch was too good to believe. I rudely pressed, "*Can I test you sometime?*"

"OK," she replied cheerfully.

Now she would eat her words...

My plan was ingeniously simple: I picked a moment when Linda least suspected. Then I boldly challenged her to name tones for me—*by ear.*

I made sure she had not been playing any music. I made her stand so she could not see the piano keyboard. I made certain that other classmates could not help her. I set everything up perfectly so I could expose her Perfect Pitch claims as a ridiculous joke.

Nervously, I plotted my testing strategy. Linda appeared serene. Then, with silent apprehension, I selected a tone to play. (She'll *never* guess F♯!)

I had barely touched the key.

"F♯," she said.

I was astonished.

I played another tone. She didn't even stop to think. *Instantly* she announced the correct pitch.

Frantically, I played more tones, skipping here and there all over the keyboard. But somehow she knew the pitch each time. She was *SO AMAZING* —she knew the tones as easily as colors!

"How in the world do you do it?" **I blurted. I was totally boggled.**

"Sing an E♭," I demanded, determined to mess her up. She sang a tone. I checked her on the keyboard. She was right on! Now I was starting to boil. I called out more tones for her to sing, trying hard to make them increasingly difficult. Still she sang each note perfectly on pitch.

I was totally boggled.

"*How in the world do you do it?*" I blurted.

"I don't know," she sighed.

And to my dismay, that was all I could get out of her!

The dazzle of Perfect Pitch hit me like a ton of bricks. My head was dizzy with disbelief. Yet from that moment on, I knew *Perfect Pitch is real.*

I couldn't figure it out...

"*How does she DO it?*" I kept asking myself. On the other hand, why can't *everyone* recognize tones by ear? It dawned on me that most musicians can't tell a simple C from a C♯, or the key of *A major* from *F major*! I thought about that. A *musician* who cannot tell tones by ear?! That's like a painter who can't recognize the rainbow of colors on his palette! It seemed odd and contradictory.

I found myself more mystified than ever. Humiliated and puzzled, I went home to work on this problem. At age 14, it was a hard nut to crack.

You can be sure I tried it myself. I would sweet-talk my three brothers and two sisters into playing tones for me, which I would then try to identify by ear. It became just a guessing game: my many attempts were dismal failures.

Next I tried playing the tones *over* and *over* in order to memorize them. I tried to feel the "highness" or "lowness" of each pitch. I tried day after day to learn and absorb those elusive tones. But nothing worked. After weeks of struggle, I still couldn't do it. Sure, Linda had an extraordinary gift—the ultimate ear for music, the master key to many talents. I wished I had an ear like that. But it was out of my reach.

So I finally gave up.

This issue only: Electronic Musician *readers get* **FREE TAPE + $50** *off!* ★

Then it happened...

It was like a miracle. A twist of fate. Like finding the lost Holy Grail...

Once I had stopped *straining* my ear, I started to listen NATURALLY. Then the incredible secret to Perfect Pitch jumped right into my lap.

I began to notice faint "colors" within the tones. Not *visual* colors, but colors of *pitch*, colors of *sound*. They had always been there. But this was the first time I had ever "let go"—and *listened*—to discover these subtle differences within the musical tones.

Soon—to my own disbelief—*I too could recognize the tones by ear!* It was simple. I could hear how F♯ sounds one way, while B♭ has a *different sound*—sort of like "hearing" red and blue.

The realization struck me: THIS IS PERFECT PITCH! This is how Bach, Beethoven, and Mozart could mentally envision their masterpieces—and know tones, chords and keys all by ear—*by tuning in to these subtle "pitch colors" within the tones.*

It was almost childish—I felt sure that *anyone* could unlock their own Perfect Pitch by learning this simple secret of "color hearing."

Bursting with excitement, I went and told my best friend, Ann (a flutist), that she too could have Perfect Pitch. She *laughed* at me.

"You have to be *born* with Perfect Pitch," she asserted. "You can't develop it."

"You don't understand what Perfect Pitch *is* or how it works," I countered. "I couldn't recognize a single note before. Now it's *easy*."

I showed her how to listen. Timidly, she confessed that she too could hear the pitch colors. With this jump start, it wasn't long before Ann had *also* acquired Perfect Pitch.

At school we became instant celebrities. Classmates would test our ears, endlessly fascinated with our "supernatural" powers. Yet to us, our hearing was nothing "super"—just natural.

Way back then, I never dreamed I would later cause a stir among college music professors. But when I got older, I eventually started to explain my discovery to the academic world.

They *laughed* at me. Many told me, "You must be *born* with Perfect Pitch; you can't develop it." I would listen politely. Then I'd reveal the simple secret—*so they could hear it for themselves.* You'd be surprised how fast they changed their tune!

As I continued with my own college studies, my "perfect ear" allowed me to progress far faster than I ever thought possible. I even *skipped over* two required courses. Perfect Pitch made *everything* easier—performing, composing, arranging, sight-reading, transposing, improvising—and it skyrocketed my *enjoyment* of music as well. I learned that music is definitely a HEARING art.

And as for Linda?

Oh yes—I'll have to backtrack...

Time eventually found me at the end of my senior year of high school. I was almost 18. In these three-and-a-half years with Perfect Pitch, my piano teacher insisted I had made ten years of progress. But still I wasn't satisfied. I needed one thing: *to beat Linda.* Now was my *final chance.*

Our local university sponsored a music festival each spring, complete with judges and awards. To my horror, they scheduled me as the last person to play—the *grand finale* of the entire event.

Linda gave her usual sterling performance. I knew she would be tough to match, let alone surpass. My turn came, and I went for it.

Slinking to the stage, I sat down and played my heart out. Guess what? I scored an A+ in the most advanced performance category.

Linda only got an A.

Sweet victory was music to my ears—mine at last!

Now it's YOUR turn!

For 18 years, thousands of musicians around the world have proved my Perfect Pitch method—plus research at two leading universities.

Now I'd like to show YOU how to experience Perfect Pitch *for yourself!* To start, you need only a few basic instructions. I've put everything I know into my **Perfect Pitch® SuperCourse**. *It's fun!* And it's *GUARANTEED* to work for YOU—regardless of your instrument, your playing style, or your current ability level.

Try it for yourself: Order your own **Perfect Pitch® SuperCourse** and play the first two cassettes. I promise you will *immediately* hear the Perfect Pitch colors that I'll start you on—*or return the Course for a full refund.* You've got my word on it. Or you can check out your progress with no risk. You'll find a *dramatic improvement in both your pitch and your playing* in only 40 days or return the Course and *I'll personally ensure you get your refund—no questions asked.*

Think of the possibilities that Perfect Pitch can open for YOU and your music. Imagine how it can improve *your* playing, *your* singing—*your own* creativity and confidence.

And picture your friends' faces as YOU name tones and chords with laser-like precision!

Please—don't *you* laugh, too! At least not until you *hear for yourself* how simple it is to discover *YOUR VERY OWN* Perfect Pitch!

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Research references at two leading universities: *A study to determine the effectiveness of the David Lucas Burge technique for development of Perfect Pitch.* M.E. Nering (1991). The University of Calgary. *An experimental investigation of the effectiveness of training on absolute pitch in adult musicians.* M.A. Rush (1989). The Ohio State University. *You will automatically receive research summaries when you order your Perfect Pitch SuperCourse or request FREE Course information!*

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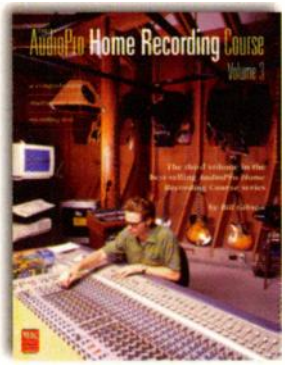

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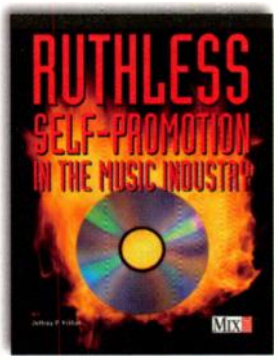

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by Jeffrey P. Fisher

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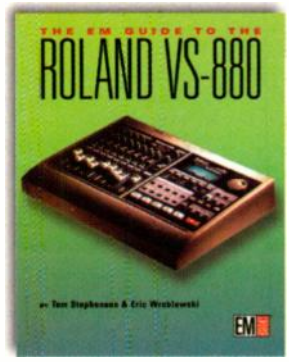

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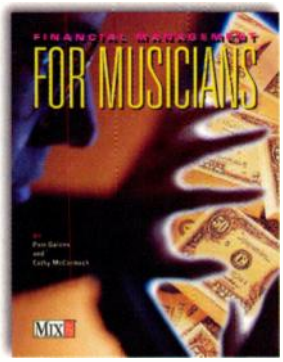

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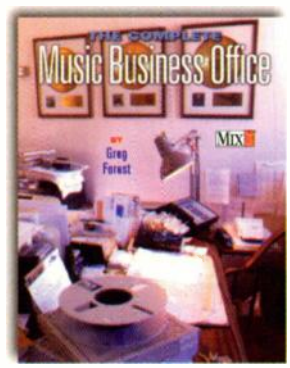

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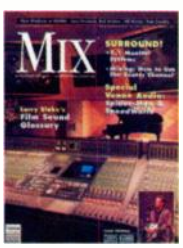

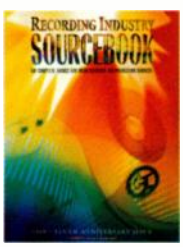

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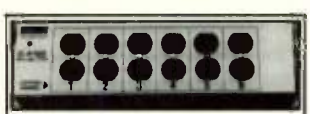

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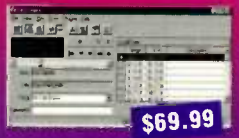

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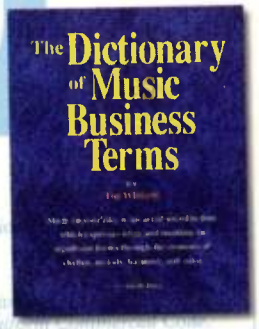

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By Larry the O

Know Thy Widgets as Thyself

At one time, the tools of audio recording were relatively few: microphones, speakers, a mixing console, a tape recorder, and a few accoutrements such as compressors and equalizers. Microphones and speakers haven't changed in shape much since then, although they have improved. The laws of physics remain pretty much the same, too.

But everything else about audio recording has changed radically, mostly due to the advent of digital technology. Computers have brought us digital reverbs and effects processors; affordable, all-digital mixing consoles with automation of all parameters; and even digital compressors and equalizers—all of which are more complicated to understand and use than are their precursors.

Additionally, new classes of tools have been introduced for audio recording and music making, such as MIDI, digital audio workstations, plug-ins, and CD burners. The greatest challenge is not, however, that things have changed and new technology has been introduced, but that the rate of change has accelerated.

Each piece of software or hardware takes time to learn fully, because today's tools are extremely feature-rich, so we are frequently more satisfied with simply employing it usefully rather than mastering it completely. Large and often poorly written manuals and badly implemented user interfaces compound the difficulties of such learning situations. How is anyone supposed to figure all of this out? I've written a lot of product reviews, and understanding a new product well enough to fairly evaluate it within a limited amount of time has become much tougher in the past five years.

If you were lucky, the most important thing that you learned in school was how to learn. With all of those digital gew-gaws to choose from, that ability is the most potent tool that you have at your disposal. If you know how to learn, you can get to first base with your gear, and you have a good chance of stealing second. I'd like to share a few guidelines I've developed to help myself quickly get things out of the box and into my work. Most of them are pretty "duh," but they *work*.

First, understand what each tool is supposed to do and what you want to do with it. If you have a surround-mixing plug-in, is it a panner, an encoder, a spatializer, or some combination thereof? Yes, you know you want to do surround mixes with it, but do you have a picture of how this plug-in fits into the overall logistics of surround mixing? Ideally, you should think all of this through before you make a purchase.

Second, figure out how to accomplish the absolute basic, bread-and-butter stuff. With a digital audio workstation, for example, the first things to figure out after connecting it are monitoring, recording, playback, importing files, bouncing to disk, and trimming regions. You'll eventually get to the fancy stuff, but the rudiments are what will help you get the work done. Read about those functions and try them to make sure you have them right.

Third, think about your own working methods and what features you need to carry them out. If you like to do your sequencing without leaving your MIDI keyboard, for example, then you need to figure out how to control your sequencer remotely with MIDI from that keyboard.

Fourth—and this is the tip that people love to hate—read the manual. Even if it's bad, it's usually better than having none at all. Although figuring out some pieces of gear is easy to do by just playing around, if the system involves a computer, the chances are good that finding and using some potentially important features will be less than obvious.

With a universe of potential before us, learning ability is an even greater key to satisfaction than is direct knowledge.

Larry the O *has been a contributor to Electronic Musician for more than 12 years, during which time he has learned his way around more hardware and software than he can remember.*

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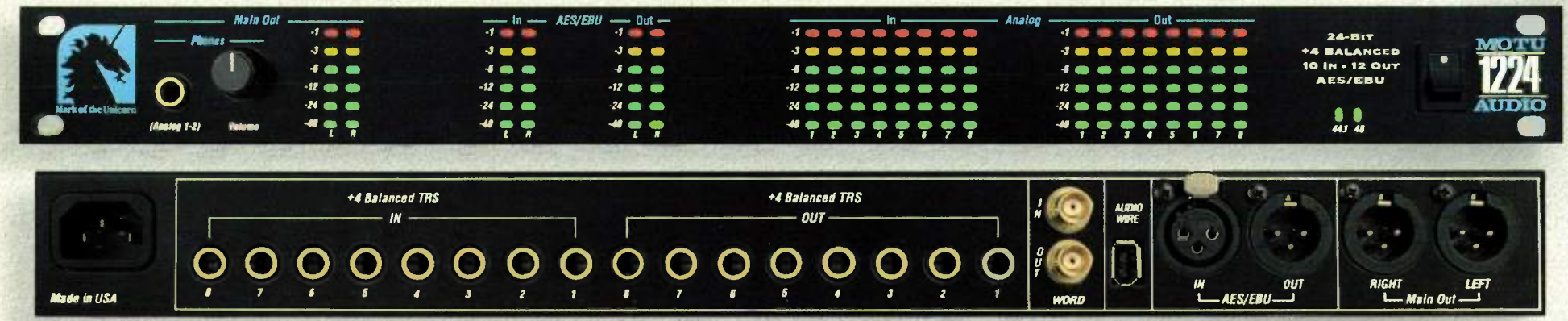

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