# Electronic Musician

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December 2001

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## OUR ANNUAL HOLIDAY GIFT GUIDE

# The Musical Palm Computer

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**Imagine walking down a beautiful red carpet filled with paparazzi and fans to accept your "just-turned" multi-platinum album. Nice dream, isn't it? Well, good news has arrived. Introducing the new VF-160 all-in-one digital power tool. It records, edits, mixes with DFX, masters, and now burns your music to a CD so you can get those hot tracks off to the record companies quicker. And forget about spending upwards of $2000! The VF-160 finally brings you everything you need at a price you can afford. So, while you may not sell sixty million records, at least you won't go bankrupt trying. The VF-160: your road to success.**

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# FEATURES

Cover photo by David Gaz

# DEPARTMENTS

# COLUMNS

*86*

*72*

# REVIEWS

# A Clear Message

**B**ack in the '60s and '70s, the federal government strictly regulated the cross-ownership of radio, television stations, and newspapers. The regulations were aimed at preventing a small number of corporations and individuals from owning multiple media outlets in the same market, thus dominating news dissemination.

In recent years, the government has loosened those laws considerably; the result is an ever-increasing consolidation of major media. It affects not only the news we get from commercial radio and TV stations but also the music we hear—and the music we aren't allowed to hear.

For an example of what could happen, we need look no further than the recent flap caused by Clear Channel Communications' ill-considered music blacklist. The list turned out to be a tempest in a teapot, but only because the company chose to weasel out of it, more or less. In case you've been living in a cave—perhaps not a bad idea nowadays—allow me to explain.

Clear Channel is the corporate owner of more than 1,170 radio stations across the United States (and 240 elsewhere), representing 110 million listeners and 20 percent of industry revenue. That makes it easily the largest owner of radio stations in the country. In the San Francisco Bay Area, where **EM** is based, the company owns two AM and five FM stations. Obviously, Clear Channel has a lot to say about what music gets played in a multitude of markets.

On September 13, some manager at Clear Channel—exactly who is not clear as of this writing—issued a list of 162 songs that the company urged its program directors not to play, on the grounds that the songs' lyrics or titles could be offensive in the current political climate. It is important to note that the program directors were *advised* but not *ordered* to respect the blacklist, and indeed, some openly disregarded it.

The list was absurd: banned were Louis Armstrong's reflective "What a Wonderful World," Simon and Garfunkel's angelic "Bridge Over Troubled Water," and John Lennon's pacifist "Imagine." But the real problem was that one megacorporation could make decisions for a large part of the U.S. radio audience.

Confronted with bad publicity, Clear Channel's top management hemmed and hawed and veered from excuse to excuse. Corporate spokespeople said that upper management had nothing to do with the list, that it was the work of a middle manager. At no time was the list retracted—at least, not publicly—but the company stated that individual program directors would be free to make their own choices.

At press time, the Clear Channel fiasco is fading. But as Grandfather said in *Peter and the Wolf,* "What if Peter had *not* caught the wolf?" Given the current social climate, had Clear Channel taken a subtler hard line and stayed with its "recommended" blacklist, it might have suffered only minor PR damage. Because its reach is so great, that policy would have had a huge impact on the music heard on commercial radio.

We have all had much to think about and do in recent weeks and months. I suggest that we keep the prevention of media consolidation high on our to-do list.

# ...INSPIRATION

## KARMA

### Technical Specifications:

- 61-key synth action keyboard with velocity and aftertouch sensitivity
- 62-note polyphony
- 32 Mbyte wave ROM, expandable to 64 Mbytes with 2 EXB-PCM expansion boards
- 640 Programs (including GM Level 2)/384 Combis – expandable to 896 Programs/640 Combis
- Support for EXB-MOSS (adds 128 new Programs/64 new Combis)
- 102 Insert/90 Master effects (up to 5 Insert, 2 Master effects plus 3-band EQ)
- 16-track 200,000 note sequencer, 200 Songs, 20 Cue Lists, 100 patterns per Song, 150 preset drum patterns, 72 RPPR patterns per Song, 16 preset/16 user Template Songs
- Joystick, 4 assignable knobs, 2 assignable switches and 1 assignable slider

### KARMA Features:

- 1190 Generated Effects (1 GE per Program, 4 GEs per Combi or Song). A GE contains over 400 parameters to generate notes, control synth and effects parameters, and provide randomization of these events
- 8 knobs, 2 switches and 2 scene memories, plus joystick, slider and pedals for real-time control over GE parameters
- 4 programmable Chord Memory buttons for triggering chord voicings easily

Appendix A: Specs **9**

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Korg USA
316 South Service Road
Melville, NY 11747

Dear Korg,

Korg products have always been terrific, but my new Karma Music Workstation is simply amazing. I continue to be blown away every time I play it. I already own a Triton, so I'm familiar with the sounds, effects and sequencer, which are great, and I like that it's compatible with all my Triton sounds.

What makes this instrument truly revolutionary is KARMA. It's brilliant! This technology is versatile, innovative and always inspires me to come up with new ideas. I'm amazed by the control that it gives me and the way I can turn a few knobs to create a completely new part. KARMA certainly is the most unique system I've seen in a long time. I produce a lot of dance music, and this keyboard continues to breathe new life into my tracks. Plus, it saves me tons of time! But I'm afraid to bring it to a live gig because someone might figure out my tricks. (ha ha)

Karma is truly the most inspiring workstation I've ever played. Thank you for creating such an outstanding instrument.

Sincerely,

Chuck Johns

Chuck Johns

STORY #23

Chuck Johns    Queens, NY

GOOD KARMA. WICKED SOUND. KORG

MIKE CRUZ/LIZABETH HEAVERN

## EXCELLENT, SMITHERS

I would like to thank Brian Smithers for his excellent review of the Tascam US-428 digital-audio interface and control surface (September 2001). However, I'll add one more thing to the stew: the US-428's six audio channels (four in, two out) are always streaming when the unit is on; as a result, the US-428 uses up 72 percent of my PC's USB bandwidth. For those who rely heavily on USB for peripherals, that means having to turn off the US-428 to use other USB-hungry devices.

I use the US-428 as a controller and MIDI interface with Steinberg's *Nuendo* and would love it if Tascam could provide a control-panel switch that would turn the audio I/O ports on and off. For people who don't use the device as an audio interface, that feature would make it a lean, mean USB machine.

**David Miles Huber**
*via e-mail*

## OURS GO TO 11

Thanks for the great magazine. I've had a subscription for only three years but have worn out most of my issues. I'm a gigging guitarist turned chemist; I'm also a home recordist because I can't give up the music bug. I

recently bought a SeaSound Solo PCI audio card and interface, which I totally dig, only to have the company go belly-up before it could release a Windows ME/2000 WDM driver [see the March 2000 issue for a review of the Sea-Sound Solo]. I took it upon myself to spearhead a user movement to gather up enough money to hire an independent firm to do it for us. Pretty grand yet stupid, eh? As Nigel Tufnel and David St. Hubbins of Spinal Tap once said, "There's a fine line between clever and stupid."

As I collected users to serve the cause, I came across Scott Mason, who has devoted a Web site (www.scottamason.com) to a SeaSound user forum for us to share ideas and opinions as well as to recruit people to accomplish our goals. We're trying to reach more users. We've posted to many newsgroups and bulletin boards about our fledgling site. Are there other avenues we have overlooked?

**Rocky Gipson**
*via e-mail*

*Rocky—Check out the company Web sites of your favorite audio-editing software. Some have forums about sound cards, including the Solo.—Steve O.*

## HISTORY IN THE MAKING

Thank you for the September 2001 issue and the two wonderful articles about Web music by Frank Jones and Eric and Karen Bell. I have never seen the process of launching a music-related Web site so concisely and effectively presented.

I wanted to create a Web site for some time but was daunted by the amount of time it would take me to understand the process. However, using those articles as models, I further researched the subtopics I didn't fully understand and, during the past three weeks, built

my first Web site—26 pages and MP3s! I credit the authors with making the process efficient and effective. This is a significant moment in my personal music history.

**Rick Toone**
*via e-mail*

## CHOCK-FULL O' GOODIES

The September issue floored me. It was full of useful information from cover to cover. I loved the section about getting your music online ("Special Delivery") and building Web sites ("Construction Site").

Also, if any one had asked me what a Foley artist was before that issue, I would not have had a clue ("Recording Musician: Sounds in the Key of Life"). What an interesting and useful field!
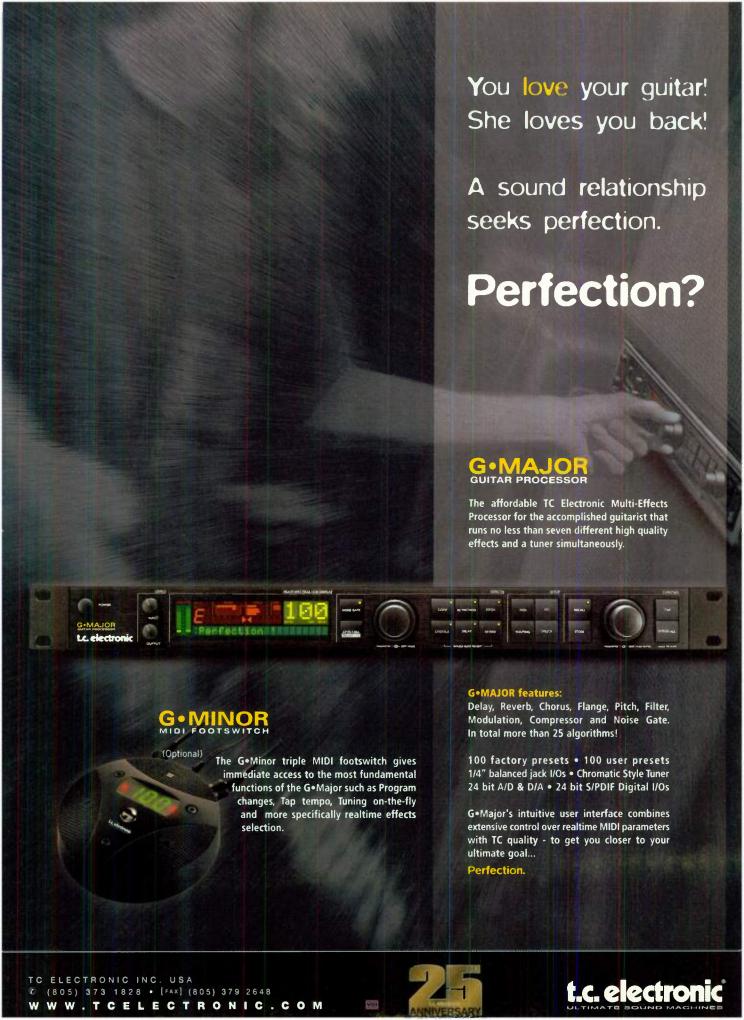
As for those who keep asking **EM** to tone down the technical talk: it's hard work, but they need to understand and relate new terms to their fields. One day they may find themselves talking with someone who deals exclusively in those terms; lack of knowledge on their part could prevent them from getting the help they need to accomplish their goals.

**Larry Tuck**
*via e-mail*

## UP IN SMOKE

In October's "Recording Musician: Off-Kilter Vocals," the round black-and-gold "yard-sale gem" shown in **Fig. 1** looks identical to a German-made Grundig I had in England in the early '60s. It came with a wonderful deco-style mono tape machine my parents bought me. Oh, if I had it now! It blew up when I tried some dastardly custom direct-wiring escapade. Keep up the good work.

**Roger Swallow**
*via e-mail*

## FIFTEEN YEARS AGO IN EM

Our December 1986 issue didn't have a cover story per se; the cover photos of a 19th-century serpent horn were supposed to visually represent sampling, showing that you could sample anything. You had to read the fine print on the table of contents to know that, but it was a cool cover.

Indeed, we ran six stories about sampling. Tim Tully (then a freelance author, later an EM editor) started the ball rolling with an overview of the market. The heart of the article was a table showing specs for 25 samplers, ranging from the $130 Casio SK-1 sampling keyboard to New England Digital's $72,000 Synclavier. Tully then explained the specs categories, giving readers a good basic understanding of how to shop for samplers. He also created a template for product roundups that we used for many years.

Next was the master himself, Craig Anderton, with one of his classic collections of useful applications ideas, tips, and tricks for using a sampler to fix problems in a mix. Michael Levine (not the Mike Levine who is the *Onstage* editor) followed with additional tips for the E-mu Emulator II, after which Steven Cox delved into the Akai S900. Anderton then returned with an explanation of how to make a sampler that appears limited to equal temperament to produce alternate scales. John Diliberto put the whole sampling suite in historical perspective with a story about sampling pioneers Pierre Schaeffer and Pierre Henry.

But we covered a lot more than sampling in December 1986. Anderton supplied an article about the state of electronic guitar, including a breakdown of what he considered right and wrong with guitar synthesizers. Helmuth Lemme also addressed guitarists' interests with a primer about guitar pickups.

The early issues occasionally covered video-oriented subjects, and Don Slepian was the resident expert. In December 1986, Slepian discussed ways of using controlled random variations to quickly generate video. In other video coverage, Matthew Leeds reviewed Electronic Arts' *Deluxe Video Construction Set* video-generation and animation software for the Commodore Amiga computer.

The December DIY section mostly focused on assorted small projects. In addition to showing you how to build a headphone monitor, continuity tester, and mono monitor (for checking mono compatibility), we offered plans for DIY security locks and other security devices for keyboards and other music gear.

That left room for just one other product review: Craig O'Donnell's evaluation of Western Automation's DASCH (Disk Acceleration/Storage Control Hardware) for the Mac. DASCH consisted of as much as 2 MB of dynamic RAM in a powered box that connected to a Mac serial port. Essentially, it provided an alternative to RAM disks, letting you run applications or the Mac OS in fast, relatively affordable D-RAM that (unlike a RAM disk) did not clear its memory when the computer was powered down. It wasn't long before RAM and hard drives increased dramatically in capacity and dropped in price, but in 1986, DASCH was cool.

—*Steve Oppenheimer*

## HOW EMBARRASSING

I tried to open a Web site (www.samplelibrary.net) that was in the July 2001 "Web Page." However, the address redirected me to a porn site! It was kind of embarrassing—I was on my parents' computer. Can you let me know if there was a mistake?

Ian Brunskill
Queensland, Australia

*Ian—There was no mistake. When I edited the July 2001 "Web Page" (during April 2001), the URL worked for the Sample Library Archive, but now it's a porn portal.*

*The biggest problem with doing a column about the Web is that sites come and go very quickly. This is the second time a site has disappeared shortly after making an appearance in "Web Page." We try hard to avoid that, but unfortunately, we can't predict how long a site will be around. It's frustrating when this happens.*

*The Sample Library Archive's new address is www.ampfea.org/sln.—Gino Robair*

## BACK AGAINST THE WALL

I have been an EM subscriber for about six years now: I love the magazine. I have a question regarding studio reference monitors.

I have a rather small control room in my home studio, and I'm having a problem with the bass. Because of limited space, I am forced to position the speakers over my mixing console at ear level, but only about three to four feet from me when my hands are on the console. At this range, I can't hear the bass well, so I am constantly moving back (with my back against the opposite wall) to hear the bass, then forward to make fader moves. Against the opposite wall, I hear considerably more bass.

Will stuffing the ports in the back of the speakers help? I have two sets of monitors, and both are a problem. I am constantly switching between my Alesis Monitor Ones and my Event 20/20s.

Jim Harris
via e-mail

*Jim—Although it's impossible to know for sure without analyzing your setup, my guess*

*is that the extra bass you hear when you move to the opposite wall is the result of a standing wave or other acoustical anomaly in your room—not an uncommon problem in home studios. Actually, three to four feet is the recommended distance for close-field monitoring, so I would trust the sound from that position more than I would from the back of the room. A helpful test would be to print two mixes, one with the bass levels adjusted according to what you hear from the near field and another based on what you hear from the back of the room, and then compare them on several other playback systems. From that you should be able to determine which is the more trustworthy monitoring position.—Brian Knave*

## A BIT ABOUT CONVERTERS

I am grateful for the wealth of information I've gained from your magazine. I have a question about recording digital audio—I am really confused. I have a 20-bit recorder (Roland VS-880EX), and I would like to upgrade to a good-quality set of standalone A/D converters (possibly the Lucid AD96).

My question is this: can I record 24-bit audio into a 20-bit machine through the digital input, or is the word length of 20 bits set for the digital inputs as well as the analog inputs? I thought the word length depended only on the converters; a friend told me that it's more than the converters and that even with standalone 24-bit converters, 4 bits would be truncated when I connect to the S/PDIF input on my 20-bit machine. Is that true?

**Jason Turetsky
via e-mail**

*Jason—Actually, you have a 16-bit machine with 20-bit converters. The VS-880EX's converters can encode audio as a 20-bit datastream, and the S/PDIF input can accept 24-bit digital audio, but the recording device is only capable of recording and storing 16-bit files. So if you were to add a set of 24-bit converters, you would still end up with 16-bit recordings. The other bits would be truncated, with dither (a type of noise) added to eliminate the resulting artifacts (such as clicks and pops).*

*That said, you still are likely to get better audio quality with good 20- or 24-bit converters than with equally good 16-bit converters, because you can run a hotter signal into the higher-bit-rate converters before digital clipping occurs. In other words, the dynamic range is greater for a 20-bit converter than for a 16-bit converter and still greater for a 24-bit converter.*

*To understand why a higher bit rate could help even when you ultimately record 16-bit files, keep in mind that a digital device does not record all signals equally; hotter signals use more bits, and softer signals use fewer bits. So a 16-bit recorder might record a soft passage using only, say, 12 or 13 bits, whereas a loud passage might use 15 or 16 bits. If you crank up the signal to compensate for the soft passages, louder passages will overload the converter, resulting in ugly digital clipping. By using a 20- or 24-bit converter, you can use a hotter signal before clipping, and you can afford to lose a few bits on the softer signals and still get a true 16-bit recording (that is, one that uses all 16 bits). That's why Roland put 20-bit converters on a 16-bit machine.*

*If you buy really good outboard 24-bit converters (they are not all equal), you could further improve the sound, not only because 24-bit converters have more dynamic range but also because the converters themselves might well be of better quality than the converters in the VS-880EX. Also, the outboard converters might have a better dithering*

*algorithm than that which Roland uses in the VS-880EX converters. If so, you can have the 24-bit converters deliver 16-bit audio via S/PDIF using the outboard unit's superior dithering.*

*Unless you actually try that yourself, you won't know whether the difference is significant enough to justify the cost of 24-bit converters. Either way, remember that the extra bits greater than 16 will not get recorded in the VS-880EX.—Steve O.*

## ERROR LOG

**September 2001, "Web Page," p. 30:** Wordlab's Web site was listed incorrectly. It is www.wordlab.com.

**October 2001, "Production Values: Cellular Electronica," p. 95:** Due to a change in Internet service providers, the Solid States Web page is now located at www.geocities.com/splattercell.

**October 2001, "Quick Picks," p. 160:** *Dan Dean Solo Winds* is distributed by Dan Dean Productions, Inc.; tel. (206) 232-6191; e-mail dandean@dandeanpromo.com; Web http://dandeanpro.com.

# WHAT'S NEW

By Marty Cutler



## SHURE KSM27

Shure's KSM27 ($575) cardioid condenser microphone offers low self-noise and a wide dynamic range. The manufacturer also claims a frequency response of 20 Hz to 20 kHz.

The KSM27 features a 1-inch ultrathin diaphragm and Class A transformerless preamp circuitry for eliminating crossover distortion. The package includes a rubber-isolated, external-locking shock-mount and a velveteen carrying pouch. Shure, Inc.; tel. (800) 25-SHURE or (847) 866-2200; e-mail info@shure.com; Web www.shure.com.

## ▶ BOSS BR-532

The Boss BR-532 ($495) 4-track portable digital studio plays sampled rhythm patterns stored in ROM and lets you record two tracks at a time. However, all four tracks and a stereo drum track can play simultaneously. The BR-532 gives you 32 virtual tracks and allows you to bounce four tracks to a stereo virtual track or to a single track.

The BR-532 uses Smart-Media cards for storage. The package includes a 32 MB card, but the BR-532 will accept cards as large as 128 MB. With a SmartMedia card reader, you can transfer your tracks to a computer for additional editing. Onboard editing features include Copy, Move, Cut, and Paste. You get one Locator point at a time and a single level of undo.

The recorder comes with two independent multi-effects processors. Effects include guitar-amp modeling, acoustic guitar



and bass simulators, delay, reverb, chorus, and EQ. You can apply effects while recording or during mixdown.

The BR-532 sports a built-in microphone, an XLR mic input, an unbalanced ¼-inch high-impedance mic input, an unbalanced ¼-inch high-impedance guitar or bass input, a pair of analog RCA inputs and outputs, a ¼-inch stereo headphone jack, an optical S/PDIF output, and a MIDI Out port. Roland Corporation U.S.; tel. (323) 890-3700; Web www.bossus.com.

## ▼ GODIN GUITARS MULTIAC JAZZ

Because of its humbucking and piezo-electric pickups, the Godin Guitars Multiac Jazz ($2,295) combines the classic hollow-body sound of an electric jazz guitar with the qualities of an acoustic instrument. At the same time, the guitar offers direct access to MIDI guitar converters and Roland virtual-guitar processors.

The Multiac Jazz supports hexaphonic string output and MIDI access from its 13-pin output jack. To attain a more traditional jazz-guitar sound, the instrument sports a humbucking pickup mounted at

the neck position. You can blend the output of the hexaphonic pickups with the humbucker to add a bit of definition to the typically mellow jazz-guitar sound. The guitar's upper bout offers a set of faders for preamp and EQ adjustments.

In addition to the 13-pin jack for the hexaphonic pickup outputs, the Multiac Jazz has one unbalanced ¼-inch output for each of the guitar's humbucking and piezoelectric pickups. La Si Do, Inc./Godin Guitars; tel. (514) 457-7977; e-mail sales@lasido.com; Web www.lasido.com or www.godinguitars.com.

# MAKE TRACKS
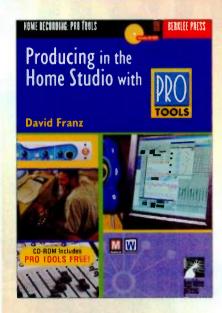
Echo's new LAYLA LapTop and MONA LapTop recording sytems easily connect to most Macintosh and Windows notebooks.

# GET SMART ▲ ▲ ▲ ▲

HOME RECORDING: PRO TOOLS                    BERKLEE PRESS

**Producing** in the
**Home Studio** with PRO TOOLS

**David Franz**

CD-ROM Includes
PRO TOOLS FREE!    M W

## ▲ BERKLEE PRESS

David Franz's *Producing in the Home Studio with Pro Tools* ($34.95) is a detailed, practical guide to recording in the personal studio. Although the book's focal point is Digidesign's suite of digital-audio recording tools, Franz covers important, related, nontechnical topics such as how to run a recording session, how to set up a studio, and recording budgets.

You also get application-specific instruction: the book provides a wealth of keystroke shortcuts and detailed technical information about Pro Tools' inner workings, including using the system's MIDI features for sequencing. Berklee Press; tel. (617) 266-2146; Web www.berkleepress.com.

## SCARECROW PRESS

Elizabeth C. Axford's *Song Sheets to Software* ($36.95) is an eclectic sourcebook for using your computer as a music-education tool. The book begins with a history of song sheets, with discussions of early American folk and religious music as well as more contemporary forms.

Ensuing chapters cover copyright laws and music software. *Song Sheets*

*to Software* finishes with a listing of more than 6,000 Web sites for music. The book concludes with a glossary of technical terms. Scarecrow Press; tel. (800) 462-6420 or (301) 459-3366; e-mail slambert@scarecrowpress.com; Web www.scarecrowpress.com.

## ▼ HAL LEONARD

The worlds of MIDI and digital-audio recording are inextricably woven together. As such, novice studio owners can find themselves awash in misinformation, misleading equipment specifications and requirements, and generally confusing advice.

Thankfully, *How MIDI Works* ($29.99) isn't another dry exegesis of MIDI protocol. Author Peter Lawrence Alexander

How **MIDI** works    tech master

guides beginners through the process of building a MIDI and digital-audio system by offering practical advice about important aspects of the modern personal studio. The book is peppered with wry observations, anecdotes, and object lessons, and it contains numerous illustrations, software screen shots, and diagrams.

Topics include audio cards and drivers, MIDI keyboards, sequencing and recording, multiport MIDI interfaces, and software synthesizers and sam-

plers. The book's chapter about understanding minimum system requirements covers everything from available audio tracks to video-card conflicts. Hal Leonard Corp.; tel. (414) 774-3630; Web www.musicdispatch.com.
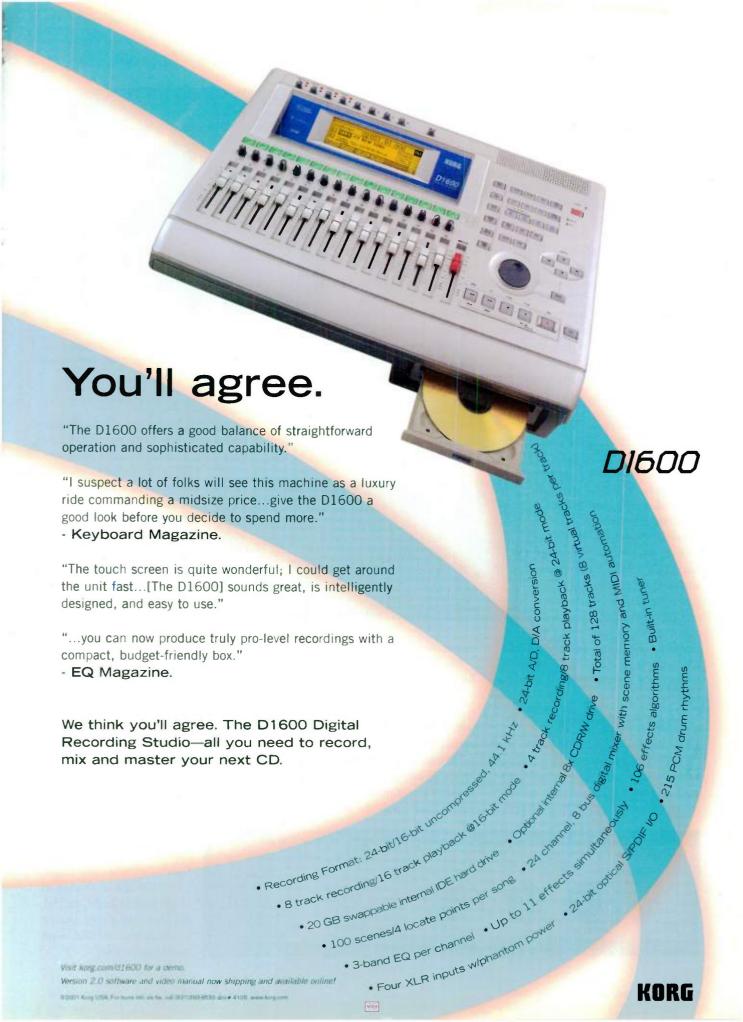
## ▼ EMEDIA

*Bass Method,* vol. 1 ($59.95), provides more than 100 bass-guitar lessons on CD-ROM for Macintosh and Windows computers. You also get more than 200 songs and exercises, all of which feature full-motion video and audio tracks. An animated fretboard helps illustrate proper fingerings as the music plays back.

The skills covered in *Bass Method,* vol. 1, range from fundamentals, such as simple bass lines and tuning, to advanced topics, including triads, fills, syncopation, and reading (including tablature and standard notation). In addition to audio tracks, the CD-ROM provides MIDI versions of the performances, which lets students vary tempos to suit their pace. The audio portion offers a bass-only track for closer inspection and a band track without bass for playing along.

Windows requirements are any Pentium, 16 MB of RAM, and Windows 95 or 2000. For Mac users, any Power PC with 16 MB of RAM and OS 7.53 or later will suffice. EMedia; tel. (888) 363-3424; e-mail custserv@emediamusic.com; Web www.emediamusic.com.

# You'll agree.

"The D1600 offers a good balance of straightforward operation and sophisticated capability."

"I suspect a lot of folks will see this machine as a luxury ride commanding a midsize price...give the D1600 a good look before you decide to spend more."
- **Keyboard Magazine.**

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"...you can now produce truly pro-level recordings with a compact, budget-friendly box."
- **EQ Magazine.**

We think you'll agree. The D1600 Digital Recording Studio—all you need to record, mix and master your next CD.

*D1600*

• 24-bit A/D, D/A conversion
• 4 track recording/8 track playback @ 24-bit mode
• Total of 128 tracks (8 virtual tracks per track)
• Optional internal 8x CDRW drive
• Recording Format: 24-bit/16-bit uncompressed, 44.1 kHz
• 8 track recording/16 track playback @16-bit mode
• 20 GB swappable internal IDE hard drive
• 24 channel, 8 bus digital mixer with scene memory and MIDI automation
• 106 effects algorithms
• 215 PCM drum rhythms
• Built-in tuner
• 100 scenes/4 locate points per song
• Up to 11 effects simultaneously
• 24-bit optical S/PDIF I/O
• 3-band EQ per channel
• Four XLR inputs w/phantom power

**KORG**

## ▶ IZOTOPE OZONE

**O**zone (Win; $199) from Izotope is a DirectX plug-in that provides a suite of mastering tools and models analog circuitry. *Ozone*'s biggest draw is its multiband dynamics processing, which includes compression, expansion, and limiting (four bands in all). A real-time graphic display illustrates the effects of processing on each band.

The Multiband Harmonic Exciter is based on analog tube-saturation models and features adjustable crossover points. A multiband parametric EQ offers soft saturation modeling and a spectrum display. A separate spectrum analyzer, with real-time and averaging modes, is also included.

*Ozone* comes with a reverb that gives you control of room size, frequency cut-off, and damping; a stereo phase meter; tools for adjusting the width of the stereo image (including multiband delays); a loudness maximizer; and an interface for determining the plug-in's signal flow and order of effects.

*Ozone* also keeps a list of all of your changes, so you can easily recall earlier adjustments. The plug-in uses 64-bit processing in its digital signal processing algorithms.

To use *Ozone*, you will need a Pentium II/ 450 MHz with 64 MB of RAM; Windows 95, 98, NT, 2000, ME, or XP; and a DirectX-compatible host application. Izotope, Inc.; e-mail izotope@izotope.com; Web www.izotope.com.



## ▶ EMAGIC EMI 2/6
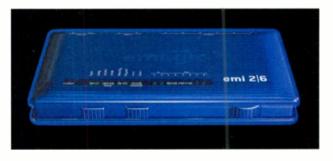
**E**magic has released the emi 2/6 ($499), a USB 24-bit digital-audio interface. The interface delivers 44.1 or 48 kHz audio with six channels of playback and two simultaneous recording channels.

Housed in translucent blue plastic, the interface provides coaxial S/PDIF I/O. The unit's two analog inputs and six outputs are unbalanced RCA connectors. The emi 2/6 is powered from a USB port and is compatible with all powered USB hubs. For situations in which you're using several USB-powered devices and running them all off of the same hub, the emi 2/6 can be powered from an external power supply (available as an option from Emagic) so that the unit



can maintain maximum audio quality.

The unit has a ⅛-inch stereo minijack for headphone monitoring and a wheel for adjusting headphone volume. Emagic claims zero-latency hardware-assisted monitoring for the interface. Additionally, Emagic offers low-latency ASIO drivers for both Mac and Windows platforms.

The emi 2/6 includes a switch for toggling between digital and analog input; another switch lets it act as the master word-clock source or as a slave to external word clock. An intelligent resampling feature allows the unit to lock to any incoming word-clock signal. Emagic USA; tel. (530) 477-1051; e-mail emagic@emagicusa.com; Web www.emagic.de.

## ▼ SUMMIT AUDIO TD-100

**S**ummit Audio's TD-100 ($495) is a half-rackspace instrument preamp and direct box. An internal high-voltage power supply drives the unit, which integrates a 12AX7a/ECC83 tube with a discrete transistor circuit at the output stage.

The input stage impedance knob is continuously variable from 10 kΩ to 2 MΩ. That lets you precisely tailor your instrument's input level. You can also plug in a set of headphones and use the TD-100 for private practice. Front-panel signal and peak LEDs allow you to monitor the input.

The TD-100 has an XLR mic-level input, an unbalanced ¼-inch input, a balanced ¼-inch line-level output, a polarity switch, and a ground-lift switch for eliminating noise from your signal. The output gain control has a range of 24 dB. Summit Audio, Inc.; tel. (831) 464-2448; e-mail sound@summitaudio.com; Web www.summitaudio.com.

# SOUND ADVICE ▲ ▲ ▲ ▲

## ▼ UEBERSCHALL

In spite of the emergence of 24-bit, 96 kHz audio, some electronic musicians never get over their fondness for the grungy, raw sound of low-fidelity samples. Ueberschall's *8 Bit Stylez* ($99.95) offers a collection of grooves, drum loops, and effects created with the Commodore C-64's notorious SID MOS 6581 chip. With the package, you get an Akai-format CD-ROM and an audio CD.

The collection offers 40 construction kits, licks, music, game effects, machine

voices, and more than 1,000 single drum and instrument sounds. Ueberschall targets the sounds for house, hip-hop, drum 'n' bass, two-step garage, and other contemporary electronic dance-music styles. East West (distributor); tel. (310) 271-6969; e-mail sales@eastwestsounds.com; Web www.soundsonline.com.

## SONIC IMPLANTS

More than 14 MB of fuzz-toned guitars constitute Sonic Implants' *Creamy Fuzz Guitar 1—Smooth Leads* (SoundFont; $19.95). The assortment includes a range of guitaristic effects, such as muted notes, harmonics, hammer-ons, and feedback. Also available are multiple-Velocity patches that switch between the sampled articulations. Sonic Implants used

a custom-designed Fender Stratocaster, a Mesa/Boogie amp, and a Fulltone Soul-Bender fuzz box to produce the samples.

One problem with creating convincing sampled distorted-guitar parts is the interaction between the strings. To that end, Sonic Implants provides *Creamy Fuzz Guitar 2* (SoundFont; $19.95). That collection includes major, minor, and suspended chords strummed up and down, as well as V7, V9, and power chords strummed downward. Sonic Implants Network; tel. (781) 641-0063; e-mail studio@sonicimplants.com; Web www.sonicimplants.com.

## ▼ GMEDIA

For its GForce *M-Tron* VST sample-playback instrument, Gmedia brings you a new compilation of sounds, *Tape Banks* ($49). The company tracked down, prepared, and recorded a number of mellotron-type instruments, including the Birotron and the Chamberlin. Each sound in a mellotron instrument uses 35 individual tapes, and each tape contains three sounds. In many cases, the tapes had to be prepared by hand for sampling.

*Tape Banks* includes the sound of wine glasses, clarinet, double bass, steel-string guitar, various choirs, Chamberlin harp, and Birotron organs. Q Up Arts (distributor); tel. (800) 454-4563

or (801) 486-8225; e-mail gforce@gmedia .com; Web www.gmedia.com/gforce.

## ▲ EASTWEST

The guitar work of studio ace Jeff "Skunk" Baxter, a veteran of Steely Dan and the Doobie Brothers, is presented on *Skunkworks* (audio and WAV, $99.95; Akai, $199.95). Baxter dishes up licks and rhythm parts on a number of guitars in the two-CD set. The styles range from acoustic-based music to jazz to heavy metal. East West; tel. (310) 271-6969; e-mail sales @eastwestsounds .com; Web www.soundsonline.com.

## BIG FISH AUDIO

Big Fish Audio's *Guitar Studio* ($99.95) is a two-CD set of guitar licks, riffs, intros, and endings that covers various styles, including alternative rock, jazz, blues, metal, and funk music. *Performance Loops Drums 2* ($99.95) is a two-CD collection of drum loops that were culled from complete performances. Included are hits, fills, and variations that will help you to generate realistic drum parts.

Both collections include a CD of audio tracks and a CD-ROM of *Acid*-compatible WAV files for your looping enjoyment. Big Fish Audio; tel. (800) 717-FISH or (818) 768-6115; e-mail info@bigfishaudio.com; Web www .bigfishaudio.com. ●

# Legends of the Small

The adage that technology develops at an accelerated rate seems all too true in the realm of microelectronics—soon to be called *nanoelectronics*. As I discussed in "Tech Page: Faster, Smaller, Better," in the October 2001 issue, IBM and Intel recently announced breakthroughs in shrinking transistors, breakthroughs that will lead to processors and communications chips far more powerful than those available today. Amazingly, the day that issue went to press, I learned of further developments that will lead to even smaller, faster, and more powerful electronic components.

Those developments are based on carbon nanotubes, in which individual carbon atoms are arranged to form hollow tubes. Scientists at Delft University of Technology in the Netherlands have succeeded in building a transistor from a single nanotube, which measures one nanometer (one billionth of a meter) in diameter—about 10,000 times thinner than a human hair. Using an atomic-force microscope, which can move individual atoms, the Delft team pushed on the nanotube, causing it to buckle in two places. Such deformities restrict the flow of electrons, letting a single electron pass through the tube at a time. That electron can be used to turn the device on or off. Conventional transistors require hundreds to millions of electrons to perform the same task.

IBM researchers also have been working to develop useful carbon-nanotube transistors. One hurdle to practical application of that technology has been reliable replication. Depending on their size and shape, nanotubes can behave as if they are metallic (conducting) or semiconducting. Previous methods of forming large numbers of nanotubes created a mixture of metallic and semiconducting types that stuck together to form ropes or bundles. Those bundles had to be separated manually to isolate the semiconducting nanotubes required to make transistors. That's fine for experimentation but far too slow and tedious for mass production.

*Carbon nanotubes*

*promise*

*single-molecule*

*electronics.*

To solve the problem, the IBM team developed a method of constructive destruction that selectively destroys the metallic nanotubes in a bundle. Mixed metallic and semiconducting bundles are deposited on a silicon-oxide wafer, and metal electrodes are formed over the nanotubes. Those electrodes "turn off" the semiconducting nanotubes, preventing any electrons from flowing through them. The metallic nanotubes are left unprotected, and applying an appropriate voltage to the wafer destroys them, leaving the semiconducting variety untouched.

IBM's ultimate goal is to create a useful electronic device out of nanotubes, and that goal is close at hand. The company recently announced that it has built a voltage inverter from a single carbon nanotube—the world's first single-molecule logic circuit. Also known as a NOT gate, a voltage inverter is one of the fundamental logic circuits that form the basis of all computers. (The others are AND and OR gates.)

To achieve that monumental feat, the IBM team created an n-type nanotube transistor, in which electrons carry the electrical current. Until now all nanotube transistors have been p-type, in which electron-deficient areas called *holes* carry the current. Voltage inverters consist of both types of transistors, and IBM found a way to convert a segment of a single nanotube from p-type to n-type, leading directly to the voltage inverter (see **Fig. 1**). More important, the inverter's gain (the ratio of output to input amplitude) is greater than one, which is essential for assembling gates into useful electronic circuits that will find their way into smaller and more powerful tools for electronic musicians.

For more information on nanotechnology, see "Tech Page: Quantum Mirage" in the May 2000 issue and "Tech Page: Nanocomputers" in the June 1996 issue. ◉
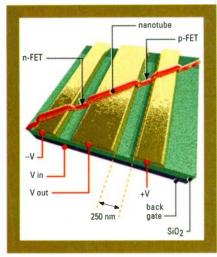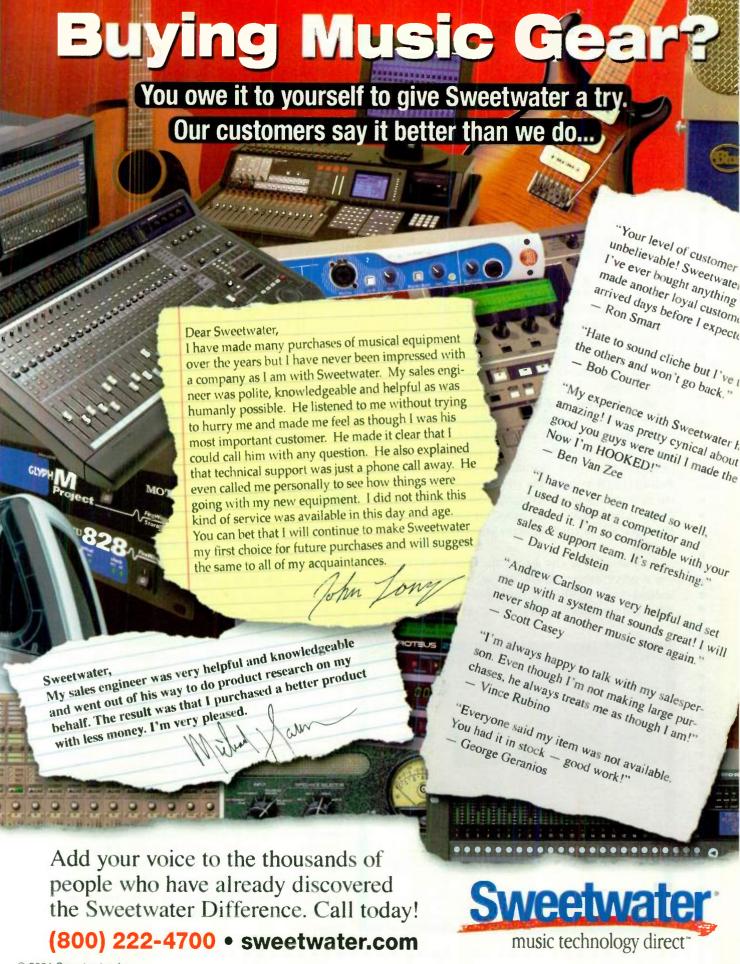


FIG. 1: In IBM's single-molecule voltage inverter, part of a carbon nanotube (in red across gold electrical contacts) is converted to an n-type transistor; the other section remains a p-type transistor. The output voltage is greater than the input voltage, which is essential for constructing useful computer circuits.

# WEB PAGE

*By Gino Robair*

## WEB SITE OF THE MONTH

If you're looking for information about American composers and musicians, check out NewMusicBox (www .newmusicbox.org). Serving as the Web magazine of the American Music Center (www.amc.net), which composer Lou Harrison once described as a "central information booth" for American music, NewMusicBox is well designed and easy to navigate—a helpful feature, because the site is rich in content.

Although NewMusicBox's topics pertain to new contemporary American music (which includes the work of deceased composers such as John Cage and Harry Partch as well as the music of living composers such as Steve Reich and the Kronos Quartet), it is far more inclusive than you might imagine. The artists covered range from the conservative to the extreme, from composer John Adams to vocalist Diamanda Galas. The curators of New-MusicBox aim to cover as many styles and genres as possible.

NewMusicBox is also a good model for a sticky site, one in which visitors will stay awhile. To begin with, it's updated regularly. The site hosts streaming audio and video, and links abound. The interview section, In the First Person, provides a text transcript (also available as an Adobe *Acrobat* PDF file) and a QuickTime movie of the interview. Audio examples of contemporary recordings in the SoundTracks sections are presented in RealAudio format. Contributing artists can bring up interesting (and sometimes controversial) topics in the section called In the Second Person, which also features a forum for visitor feedback. Recent topics include "Your Career or Your Life" by John Luther Adams, "Finding Your Voice" by Keith Corbin, and "The World Trade Center Tragedy and New Music" by NewMusicBox editor Frank J. Oteri.

You also get current events in the News section, composer and performer perspectives from In the Third Person, a concert calendar of American works in Hear and Now, and Hymn and Fuguing Tune, in which a group of artists are asked to respond to a question. Rather than serving as a source of jingoistic pride, NewMusicBox provides a resource for information about creative music emanating from around the country.

## DOTDOTDOT.COM

The **Freesound** mailing list focuses on freeware, shareware, and topics related to unusual sound-design software and techniques. Some of the programs discussed are Tom Erbe's *SoundHack* (see the July 2000 "Web Page"), Audio Ease's *Thonk,* and Phil Burk's Java-based application program interface, JSyn (see the June 2000 "Web Page"). You can join the Freesound list by sending an e-mail to majordomo@shoko.calarts.edu with "subscribe freesound" or "subscribe freesound-digest" in the body of the message. . . . Owners of vintage Korg MS- and PS-series analog synthesizers should take a look at Benjamin Ward's **Korg Kornukopia** (www.benward.net/korg). The site offers transcriptions of the user and service manuals (complete with typographic and technical mistakes), user tips and techniques, and fan-related items such as brochures and photos of the wonderful late '70s instruments. . . . An interesting and reliable source for older electronic instruments is the **Vintage Electric Musical Instrument Auctions** (VEMIA; www .vemia.co.uk). Located in the United Kingdom and administered by Peter Forrest, author of the *A–Z of Analogue Synthesizers,* VEMIA's auction list typically includes synthesizers, amplifiers, microphones, effects, and drum machines as well as odd and unusual items. Well-known customers of the site include Brian Eno and members of Oasis.

# WEBCAST

Certainly, one of the most outstanding sites for classical-music fans is Andante (www.andante.com). The site's creators hope that Andante will become *the* online source of information for classical-music enthusiasts, and so far they are headed in the right direction.

Andante provides eight general areas to explore: Musicroom, Magazine, Calendar, Directories, Reference, Profiles, Learning, and Boutique. To hear music, head directly to the Musicroom, where you can link into Andante radio for streaming audio or view exclusive Webcasts of important performances. For news and information, Magazine and Calendar offer up-to-date listings and links. The Magazine area includes relevant news, interviews, and concert and CD reviews. The Calendar section covers hundreds of concert listings worldwide and includes a search engine for searching by date, key word, country, and venue.

Directories offers links to no less than 12,000 Web sites dedicated to classical music. The links are grouped by categories such as Composers, Ensembles, Festivals, and New Music Links. That section alone is worth bookmarking.

From the Reference section, you have access to the *Concise Grove Dictionary of Music,* the *New York Review of Books,* and *New Kobbé's Opera Book,* among other resources. Included under the Profiles heading are previews of upcoming projects, such as Andante's planned partnership with IRCAM, the renowned research center in the Centre Georges-Pompidou in Paris, France.

Boutique features Andante's reissues of rare and historically important recordings. Although still in the works, the Learning section will eventually include online educational resources.

Unlike other venues for classical music, Andante is not afraid to cover contemporary composers and performers. For example, Pierre Boulez, known as much for his interpretations of major symphonic works as for his own challenging compositions, was the featured artist on the site at the time of this writing. Whether you're a classical-music buff or just getting your feet wet in the genre, Andante is an important destination for your browser.

# DOWNLOAD OF THE MONTH

More than just a source of free samples, AnalogueSamples .com (www.analoguesamples.com) is a community of synthesizer enthusiasts willing to share its sonic discoveries with others. Launched in 1999 by Stefan Walter and Björn Fogelberg, both of Sweden, AnalogueSamples.com contains thousands of interesting sounds.

The samples are grouped into 14 categories, including drums, leads, keys, pads, brass, and bass. Once you choose a category, the sounds are presented in a list, with useful information about each sample: name, rating (based on votes from site visitors), number of files available, type of synth used, description of the sound, and file size. A RealAudio version of each sample is provided, so you can preview the sound before you download it.

On AnalogueSamples.com, you can just as easily view samples by synth type. If you have ever wondered what kind of sounds a Serge modular can make, this site will let you hear one. Simply scroll down the long menu of synthesizers until you find one you want. The list includes common instruments (such as the ARP Odyssey

and Oberheim Xpander) and rare ones (Buchla 200-series and E-mu modulars). Besides a link to related samples, each synth page includes a photo, a brief description of the instrument, and a link to a related site if available. The site also lets you rate the samples for the synth you chose.

Rounding out AnalogueSamples.com are a few other notable pages. Documentation for older synths (including Akai, Korg, Moog, Roland, and Yamaha) and popular chips (CEM 3340 and SSM 2044, for example) can be located among the Manuals pages. At the Store, you can purchase a CD-ROM of Akai-format samples called *Killer Tweaks—Analog FX,* made using Roland System-100 and SH-09 synths.

To use AnalogueSamples.com, you must register, but registration is free. If you're interested in checking out even more sounds, for a $10 monthly fee, AnalogueSamples.com gives you FTP access to additional samples as well as its entire site.

## 🌐 BAND ON THE WEB

Arriving from the frog planet Kero Kero are the three musicians who form eX-Girl (http://0505.net/ex-girl). Their mission: to save the human race with Kero Kero music, which features unearthly vocals and odd, unpredictable songs. (The word *kero* is the Japanese equivalent of the frog utterance *ribbit.*) *Back to the Mono Kero* is the band's fourth CD and its U.S.-label debut on Ipecac Recordings (www.ipecac.com).

Beyond the sci-fi trappings, eX-Girl, at first blush, comes off as a Japanese neo-power-pop trio. But underlying eX-Girl's overt cuteness are staggering vocal parts, clever writing, intricate arrangements, and inspiring musicianship worthy of heavyweights such as Frank Zappa, the Art Bears, and Captain Beefheart's Magic Band.

The trio demonstrates considerable vocal prowess with complex and demanding parts that exploit the dynamic range of the members' voices. For example, eX-Girl may begin a song singing standard rock vocal parts, dissolve into Bulgarian-inspired close harmonies, and then drop into an overdubbed operatic chorus reminiscent of Queen's "Bohemian Rhapsody." Collaborations with progressive-rock legends Gong, Magma,

Hugh Hopper, and Chris Cutler are a testimony of eX-Girls' musicianship.

It is surprising, then, to hear that the members of eX-Girl picked up their instruments for the first time four years ago, when they first formed the band. "They were just singers before forming eX-Girl," says the band's producer and chief collaborator, Hoppy Kamiyama (www22.cds.ne.jp/~inochi/kamiyamania/findex.html). "Kirilo, the leader of eX-Girl, chose the instruments for each member using her sixth sense."

Kirilo's sense was spot on. EX-Girl's guitarist, Chihiro, is as agile at playing rhythmic power chords as she is at twisted melodic lines. Fuzuki has the strength and focus of a Taiko drummer while standing and playing only a floor tom, snare drum, and some cymbals. Kirilo—whose label, Kiki Poo Records (www.kikipoorecords.com), is the home of previous eX-Girl releases—gave herself bass duties. Live, she also plays a small Casio with her foot.

Kamiyama, an internationally renowned musician and producer who has worked with notables such as John Zorn and Marc Ribot, has produced and shared songwriting credits with the band since the beginning. In eX-Girl mythology, he is known as the Frog King. "Outer space is the Utopia where eX-Girl has forever lived with Frog King," Kamiyama says. "I tell them anytime I get a great idea from the Frog God of the entire universe. Then, we realize that we are creating these great songs by means of a fantastic telepathy from planet Kero Kero."

Often the messages come in the form of English lyrics that are delivered in a manner that playfully confuses their meaning. In "Cucumber Surrender," Kirilo delivers her lyrics in a quasi-operatic style that gives its surreal innocence a bit of overt sensuality: "Before you slice him up and make him a meal / Stroke your cucumber / Let him know how you feel / Lettuce bed him with Princess Jalepino / He saved the world of vegetables we know."

Another item that stands out on *Back to the Mono Kero* is eX-Girl's cover of Robin Scott's dance hit, "Pop Musik." The pedestrian four-on-the-floor beat is pushed along by the tight chorus punctuations and dozens of intricate interjections by the vocalists.

The songs on *Back to the Mono Kero* are rife with so many melodic twists and turns that you may think they result from hours of studio manipulation. However, eX-Girl is adept at delivering its focused performances live and in the studio. "I never edit the songs," Kamiyama says. "It was kind of like making a live recording. I only spent seven days recording and mixing *Back to the Mono Kero.*"

When asked which digital audio workstation he uses to make records, Kamiyama revealed his preference for older technology. "I've never used a computer system to record eX-Girl," he says. "I use a 16-track analog recorder with ½-inch tape. I just sprinkle eX-Girl's sound with strawberry magic called gram pot. That's true, indeed!" ☻



SAGUARO

# I'M BIASED

## Morris Hayes
**Programmer & Keyboardist with Prince & The New Power Generation**
**Producer for Prince's New Power sampling library**

*Photographed by Mike Woodside at Paisley Park Studios, Minneapolis*

**T**here is an intersection between technology and art. It's where things happen. It's a proving ground. It's where sonic creativity flows on a path of least resistance. At BIAS, it's where we put our energy. Chances are, it's where you put yours.

Morris Hayes knows that intersection well. As a veteran band member for Prince, and now as the producer for Prince's New Power sampling library, Morris depends upon BIAS audio software products to keep his own sonic creativity flowing: "BIAS gives me what I need — to do what I do. Anything I can hear in my head, BIAS software lets me turn into reality. Simple as that."

By Julian Colbeck

# Moog Music Prodigy

*Produced: 1979–1984*
*Made in: United States*
*Designed by: Rich Walborn and Tony Marchese*
*Number produced: 11,000*
*Synthesis system: analog, subtractive*
*Price new: $645*

| *Today's prices:* | *Like new* | *$700* |
|---|---|---|
| | *Like, it's okay for its age* | *$500* |
| | *Like hell* | *$300* |

Because Moog Music created the Prodigy without any assistance from the company's namesake and founder, Robert Moog, the Prodigy inspires heated debate about the legitimacy of its pedigree. Is the Prodigy as authentic a Moog as the classic Minimoog, or is it an imposter?

Today the Prodigy is in such demand that prices for models in great condition actually top the original price (inflation notwithstanding). Nonetheless, some claim that it's an insipid shadow of a "real" Moog. A quick dive into its history might reveal just how authentic and valuable the Prodigy really is.

The Prodigy was conceived as a cut-down instrument; its creators never intended it to rival its elder siblings. A team of Moog employees, including designers Rich Walborn and Tony Marchese and project manager Tom Rhea, brought the Prodigy to life in an attempt to build a dual-oscillator synthesizer for about $500. Their success was, in part, a result of design innovations such as potentiometers mounted on the circuit board rather than bolted to the front panel as they had been on the Minimoog. Placing the final loudness and filter-cutoff controls in the middle of the front panel was another inventive departure from previous Moog instruments.

The Prodigy looks quite striking, with real wooden end panels and an unfettered control panel. Its informative front-panel graphics make the Prodigy a perfect setting for learning about subtractive synthesis. Aside from the occasional missing knob or switch,

it's not altogether uncommon to find one in nearly perfect condition.

Either of the two voltage-controlled oscillators (VCOs) has a switch to select sawtooth, triangle, or pulse waveforms; Osc 1's pulse wave is narrow, and Osc 2's is square. The oscillators can be tuned within a two-octave range; Osc 1 is calibrated from 32 feet to 8 feet, and Osc 2 from 16 feet to 4 feet. An interval knob can split the voices as much as a fifth apart. A sync switch lashes them back together for those searing, forced tones that remain de rigueur in the Jan Hammer school of lead synth playing. A small mixing panel offers a master volume control and independent control of each oscillator's level.

The Prodigy doesn't implement the "heated chip" technology of some other Moogs, so its pitch stability can sometimes be less than reliable. If you're thinking of buying a Prodigy, you'd be wise to leave it turned on for a few minutes to reveal whether its oscillators are subject to drift.

The 24 dB–per–octave lowpass filter features the usual cutoff frequency and resonance (which Moog persisted in calling *emphasis*) controls. Like the amplifier, the filter is
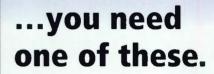


One of the most sought-after monosynths on the market, the Prodigy isn't quite in the same league as the Minimoog. With the right name and some of the right circuitry, though, it's clearly an acceptable alternative.

# The Perfect Pro Tools® Companion: TASCAM MX-2424

**If you have one of these...**

**PRO TOOLS**

**...you need one of these.**

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

Why is the **TASCAM MX-2424** the perfect companion to your Pro Tools or other DAW system? One word: *compatibility*. The MX-2424 offers your choice of two native audio file formats: Sound Designer II on Macintosh-formatted drives, and Broadcast Wave on PC formatted drives. These files support time stamping, giving you a fast, convenient way of transferring audio into your Pro Tools or other DAW system that supports time stamped audio files. Instead of spending hours aligning each track to its approximate original location, your recordings will be where they belong with sample accuracy.

Other reasons to get an MX-2424 for your Pro Tools rig? Since the MX-2424 records to SCSI drives, you can hot-swap them between systems without powering down your computer and recorder. You can record remote performances conveniently, leaving your computer in the studio. You can enjoy a familiar interface with the classic feel of a tape recorder. And perhaps most important, you can bet that the MX-2424 will satisfy your highest expectations in audio quality. If your editing/mixing system is based around Pro Tools or any other DAW, check out the ultimate companion piece – the MX-2424 – at your TASCAM dealer today.
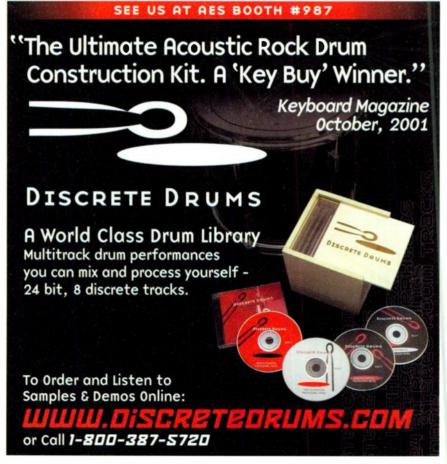
*If you don't have a DAW system, TASCAM's MX-View waveform editing software runs in native Mac and Windows versions and connects via a fast 100Mb Ethernet interface. With MX-View, you get sophisticated, sample-level waveform editing, drag-and-drop editing on the fly, click and pop repair with the pencil tool, onscreen metering for up to six MX-2424s, editing across multiple machines, easy management of virtual tracks and much more.*

*For all the details on the MX-2424 go to*
## www.mx2424.com
All copyrights are the property of their respective holders

# TASCAM.
## a whole world of recording

modulated by a slightly limiting attack-decay-sustain (ADS) envelope generator (which Moog called a *contour generator*). The filter can also track the 32-note keyboard with three settings—off, half, and full—which is good for playing self-oscillating whistles. The low-frequency oscillator (LFO) offers only square and sine waves to modulate the voltage-controlled filter (VCF), VCOs, or both.

Performance controls include portamento and rather stiff Pitch and Mod wheels (though much less stiff than they once were). The designers could have included many additional features, but only by increasing the Prodigy's cost. The spin-off benefit to such a no-frills approach is that nobody is ever intimidated with bells and whistles such as sample and hold or a multitude of modulation sources and destinations.
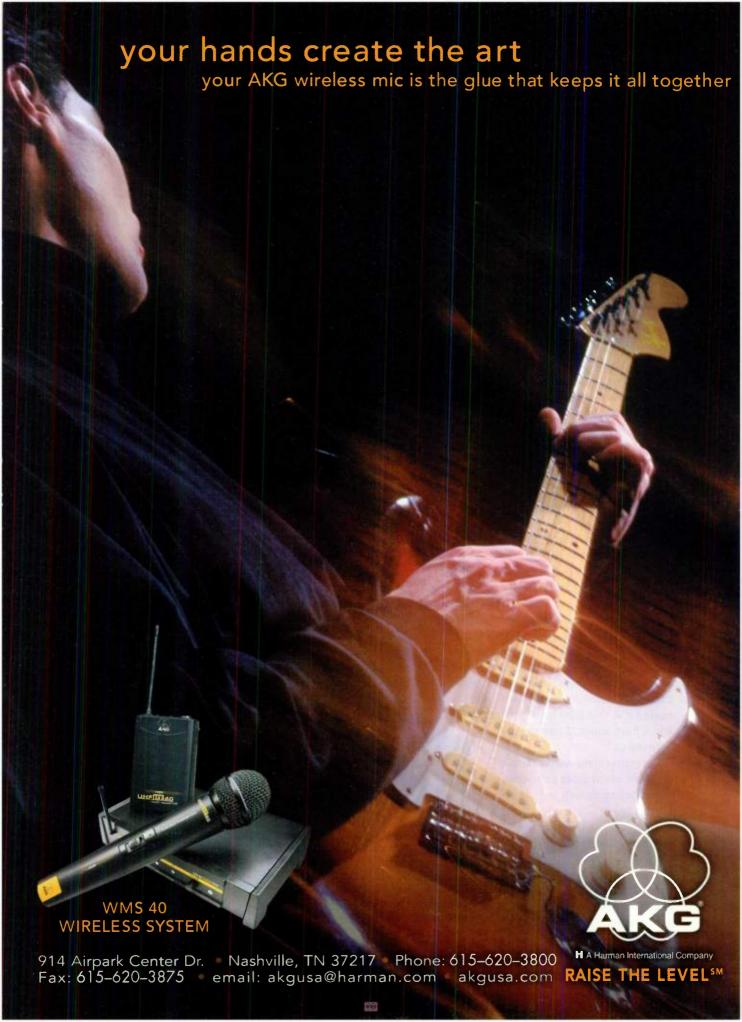
The Prodigy isn't particularly flexible, but it does conjure the basic sound of a Moog synthesizer. Limitations can be a good thing. It is extremely easy to coax all manner of sounds, from screaming leads to fat bass, from the Prodigy. Its hard-sync oscillator timbres are especially good and have contributed greatly to its popularity.

The Web offers plenty of fairly useless sites on the Prodigy. One useful page (www.teknospace.com/prodigyout .jpg) offers an image of the Prodigy's control panel, which you can print out and use to record your patches.

The affordable-yet-powerful Prodigy is a child of the 1980s. It was snapped up by performers such as Howard Jones, 808 State, and Depeche Mode. The Prodigy's initial success was greater in bargain-conscious Europe (where it effectively obliterated England's Wasp monosynth). But in the end, it became Moog's biggest seller next to the Minimoog.

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**Julian Colbeck** *has toured everywhere from Tokyo to São Paulo with artists as varied as Yes, Steve Hackett, John Miles, and Charlie. Special thanks to Tom Rhea of Berklee College of Music.*

# Forbidden Planet

*Neil Haverstick discovers other worlds of sound within a microtonal universe.*

Although guitarist Neil Haverstick lives in the shadow of the snow-covered Rocky Mountains, he might as well be from another planet. A devotee of *microtonal music*—essentially, sounds that incorporate more semitones than the standard Western scale of 12 tones per octave—Haverstick uses modified guitars to conjure unsettling waves of color that wouldn't sound out of place in a science-fiction movie.

"Once people understand that there's something called microtonal music, that's half the battle," says Haverstick, who resides in Lakewood, Colorado. "When I explain it to people, they say, 'Far out.'"

Haverstick used custom guitars—a 19-tone Starr, a 34-tone Fender Telecaster, a fretless Roland GR-303, and a Korg X911 guitar synth—to record his third album, *Other Worlds*. Recorded mainly in bare-bones studios, *Other Worlds* captures the tones common to gamelan and Indonesian music but with textures that sound positively extraterrestrial. "That's what I enjoy the most: not doing the same thing over and over again," Haverstick says.

Friend and instrument maker John Starrett introduced Haverstick to the strange hybrids, which proved to be an eye-opening experience. "He handed [a microtonal guitar] to me and said, 'Mess with this,'" Haverstick says. "At the time, I was familiar with Ravi Shankar but hadn't really soaked it in. One thing led to another, and now [Starrett and I] have about 20 instruments between us."

The guitars come with scales all their own, which poses a challenge for recording as well as for performing live. Trial and error taught Haverstick each instrument's unique abilities. "I tuned to the harmonics and simply tried to find the notes that sounded good," Haverstick says. "A lot of our guitars are refrets. For the most part, I tune them normally, but I'm not against tuning them differently."

Despite the science fiction references on *Other Worlds,* Haverstick takes a decidedly lo-fi tack when it comes to recording. "I am about as low-tech as you can get on this planet," Haverstick says. "For me, that's part of the charm—not knowing what [technology] is being used. I'm only going to learn as much as I have to. As we age, our energies need to be judiciously allocated."

*Other Worlds* contains four tracks. The title track was recorded live at the Microstock 4 microtonal music festival in September 1998 on a rented DAT machine and edited in Sonic Foundry's *Sound Forge*. In 1999, at friend Jim Ratts's Raven Studios, Haverstick tracked "The Spider" and "Didgeridon't" from his Korg A1 effects processor straight into an 8-track Fostex E-8, which uses ¼-inch tape. He recorded the album's final track, "Nebulae," in 1988 on his Fostex 260 4-track cassette recorder. Even the effects on his guitars border on minimal. Haverstick uses the A1, an Alesis MicroVerb, and Boss and DigiTech delays for some tracks and no effects at all for other songs. "The Korg A1 is a monster effects box," Haverstick says. "The parameters in this thing—you can adjust them until you're ill."

Haverstick is working on his fourth album, *If the Earth Was a Woman,* and is starting to use his computer for recording music. Although he believes microtonal music holds the possibility of rejuvenating the music world, he admits Western ears may not be attuned to the sounds common in Indonesian and gamelan music.

"I want to kick my fellow musicians in the rear," Haverstick says. "Let's do stuff that hasn't been done before." ☻

*For more information, contact Neil Haverstick; PO Box 150271, Lakewood, CO 80215; e-mail stick@ uswest.net.*

*Other Worlds*/Neil Haverstick
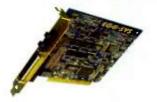
NEELA HAVERSTICK

**A**s a child, I loved gadgets: secret decoder rings, bat utility belts, all those sorts of things. So it figures that, as an adult, I have become a Palm addict. Desktop computers simply aren't enough; I need a cool small device that fits in a purse or a pocket and does an amazing array of gadgety things. As you might expect, many people share my addiction.

Although many people know about the Palm handheld computer's prowess at storing and organizing phone numbers and appointments, they might be surprised by the variety of music-related hardware and software products for the Palm OS platform. I will discuss some of the most interesting products, but for readers unfamiliar with the Palm computer, here's a brief introduction.
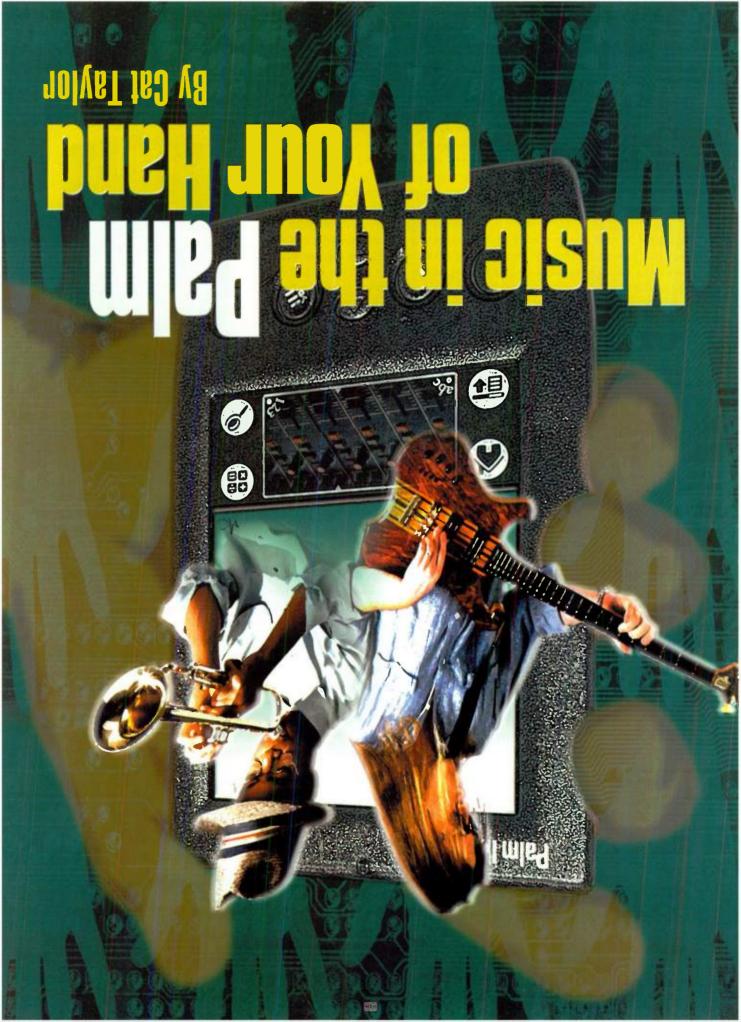
## Explore the Palm handheld computer's amazing world of music-making hardware and software.

### GETTING STARTED

Those who haven't used a Palm Personal Digital Assistant (PDA) need to know a few basics. You enter information into the Palm by writing and tapping on the screen with a *stylus*—also called a pen—using Graffiti, the Palm's slightly awkward alphabetical entry system. You install software and synchronize files between the Palm and a PC or a Mac by placing the Palm into a cradle that mates with its bottom-mounted serial port. A cable connects the cradle to the Mac or PC serial or USB port. The software that controls the synchronization process is called the *HotSync Manager*.

The Palm is a handy little contraption, but what can it offer musicians—especially considering that the Palm lacks a true speaker or amplifier? (Its rudimentary sound output is produced by a piezoelectric element.) Even when the software lets you reformat WAV files as Palm database (PDB) files, the speaker reproduces sounds inadequately. The Palm OS processes MIDI data, but only Format 0 Standard MIDI Files (SMFs) are supported, so you can't edit tracks separately.

Illustration by Mike Cruz

# Music in the Palm of Your Hand

None of those facts deter programmers, who continue to develop music-related programs at an amazing rate—I soon realized that I couldn't review all of the products on the market. I therefore focused on hardware and software that is compatible with the Palm III series of handheld devices and intended for real musical applications, as opposed to the many tools that let you catalog MP3 files, record thoughts using outside hardware, or play movies (see **Fig. 1**).

Many programs not intended strictly for musicians can prove to be valuable as well. For example, some programs let you convert Microsoft *Word* documents and PDF files into miniature versions that can be synced to the Palm. The files can include everything from documents about basic scale forms to orchestration databases created for TealPoint Software's *TealInfo* database program. Once you own a reader program, you can download a lot of free material from Web sites such as Memoware.com, which boasts more than 8,500 titles in various Palm formats. I wrote much of



**FIG. 1: The Palm series of handheld devices has spawned a surprising number of music-related applications from studio-equipment controllers to notation programs.**

my initial research for this article using the Palm Portable Keyboard and DataViz *Documents to Go*.

## TEACHING TOOLS

I'll start with some relatively simple programs, such as Sam Trenholme's freeware *Ear Trainer* (see **Fig. 2**). The program plays a random interval; you guess the interval, and it tells you if you're correct. That's all there is to it.

Wilson Cheng's *MusicEar* is a suite of ear-training programs that includes modules for learning intervals, chords, scales, rhythms, and melodies at increasing levels of difficulty. Once I figured out how to answer the questions, it was fun and challenging. *MusicEar* could be a great tool for music students and practicing musicians who want to brush up on their skills.

Mike McCollister's *McChords* lets you select more than 70 chord or scale names that are then illustrated on a keyboard and played back. It's helpful for understanding what an Am6(9) chord or what a pentatonic minor scale sounds like.

Programs designed to help guitarists determine chord fingerings are also available, but I had to draw the line somewhere.

## KEEPING TIME

The Palm OS uses a 10 ms clock, and the timing of programs using an incompatible scheme could be uneven. Unless a program has a checkbox for "ignore system information" or something similar, it's possible that the timing can be thrown off by hacks, such as the one for the Palm keyboard that lets you type text into the Palm. *Hacks* are small programs that allow changes within the basic Palm system to enable everything from fancier interfaces, such as Synergy Solutions' *Launch 'Em,* to fun things, like *Star Trek* sounds. Managed by programs such as DaggerWare's *HackMaster,* hacks can be turned on and off. If you're having problems with irregular rhythms, especially with a program as labor intensive as miniMusic's *BeatPad,* try disabling your hacks.
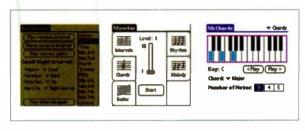


**FIG. 2: Several Palm programs can help improve your music skills, including (left to right) Sam Trenholme's *Ear Trainer*, Wilson Cheng's *MusicEar*, and Mike McCollister's *McChords*.**

Responsive Software's *Responsive Metronome* is a good basic metronome that uses visual cues (a note moves from side to side) or sound to provide a basic tempo (see **Fig. 3**). A slider along the left adjusts the tempo.

*TS Tempo* (included with notation program *TS Noter*), from Italian developer TobelStudio, is another decent metronome that features an old-fashioned wand moving from side to side. It includes a bpm setting and lets you choose sound and visual cues.

Eric Cheng's *Metronome* comes with *PocketSynth*. It also offers a bpm feature with a dot that beats on the left for downbeats and on the right for subsequent beats. If you tap the Scroll Up button repeatedly, the program measures the tempo you're tapping.

Gary Duke's *Pocket Beat* has an animated character sitting at a drum set. You can set up two tempos (each with a shuffle option) and switch quickly between them. When you hit the Go button, a little spot flashes on the kick and snare drums as the corresponding synthesized sound plays. You can also tap a count off to set the tempo. Although the program developers claim that it's the most accurate Palm metronome, the squeaky sound and size of the program (you must install *PocketC*) are drawbacks.

Calvin Gaisford's *Musician Tools* is a suite of nifty programs that includes a tuner, a metronome, and a circle-of-fifths program. The tuner lets you choose the base frequency, so you can tune to A-442 if you want. You can control the length of time that the note plays, and you can choose other pitches. The circle-of-fifths program lets you go around a circle of notes, and as you tap

# THE NEW DIGITAL WORKSTATION
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nominated big brother, the AW4416. In terms of features and performance they're remarkably similar. So, the AW2816's price –
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- 4-Band parametric EQ and dynamics on all channels, all the time**
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### CREATING 'KANDO' TOGETHER

www.AW2816.com

each one, it shows you the sharps or flats for that key. *Musician Tools'* metronome is the best of the bunch. The downbeat sounds a pitch different from the other beats, and a segmented vertical indicator shows the current beat, which helps if you want to turn the sound off. Moreover, you can choose meters with as many as eight beats to a measure.

## TECH TOOLS

The Palm can serve as a valuable portable reference resource to help solve studio-related problems, wade through technical jargon, and make calculations. Palmetto Logic's *Sound Advice*, the most complete product in this category, rolls those capabilities into a single three-part package (see **Fig. 4**).

In *Sound Advice*'s Q&A View section, a question field at the top lets you choose a general question, such as "How do I avoid feedback?" The program then offers a list of related topics, such as "feedback, notch filter," and provides specific information about them. Choosing All in the question field shows all the topics, and you can bookmark single entries for later reference. When you click on a topic, it puts you in Advice view; from there, the forward and back arrows take you to related subjects. Clicking on the Note button creates a dated attached note that you can beam to other *Sound Advice* users with the Palm's infrared port.

The terminology section contains an alphabetical list of more than 200 terms. You can access terms by scrolling down the entire list or by tapping in the Index pop-up box to jump to a letter; selecting a term displays its definition. The Reference Charts section includes a chart for combined speaker
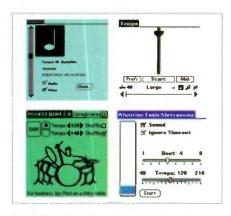


**FIG. 3: With the right software, you can transform your Palm device into a full-featured metronome. Shown are (clockwise from top left) Responsive Software's *Responsive Metronome*, TobelStudio's *TS Tempo*, Calvin Gaisford's *Musician Tools*, and Gary Duke's *Pocket Beat*.**

impedance, recommended cable lengths, common value multipliers, and musical-scale frequencies. It also provides two calculators, Volt/Ohm/Amp/Watt and RMS/Avg/Peak, that allow you to make

calculations relating to current and voltage, current and resistance, voltage and resistance, and power and voltage.

*Sound Advice* demands a lot of room for a Palm program—186 KB—but it has quite a bit to offer. At $7.95, it's a great deal.

## KNOW THE PLAYERS

*Palm Tracker,* from German developer Emperor Studios, is a tricky program that seems to play three music tracks at once by slightly delaying each track relative to the previous one (see **Fig. 5**). You create patterns using a small keyboard on the Patterns page and then arrange the patterns on the Song Editor page. Its rather simple note-entry capability only allows for short note values that are combined to sound like longer ones. You can't really change the basic 4/4 meter and feel, and the songs are not MIDI compatible.

Although you hear the songs through the Palm's speaker, they sound kind of cool. *Palm Tracker* has been used by game developers to create music for games such as *Archon* from Free Fall Associates. The full version lets you save files that you can export to your desktop computer and share with other users, although you can't open the files with desktop applications.

MiniMusic *BeatPad* is a pattern-based sequencer that allows you to create a repeating pattern of melody notes (and drum patterns if you're using MIDI). It allows you to make changes as the pattern plays and provides interactive and graphical control over pattern parameters, including the pitch, duration,

and volume of each note as well as the pattern's length and tempo.

*BeatPad* provides two main Views: Melodic and Drum. Across the top of the Melodic View window are the controls for the 32 patterns you can create and use. Below that, the Pattern Overview acts as a viewer for the pattern creation process. Four tabs let you choose and edit pitch, duration, Velocity, and master, the last of which controls the master settings that don't

change when you switch patterns, such as the tempo and the MIDI instrument patches chosen for the melodic and drum lines.

You can switch to the Drum View as a sequence plays by tapping on a button in the lower left of the screen. Because the Palm cannot deal with multiple sounds, drum sounds are audible only when using a MIDI interface. The drum screen shows the percussion sounds, lets you solo or mute them, and allows

FIG. 4: Palmetto Logic's *Sound Advice* packs a wealth of useful reference resources and studio tools into a single package.

FIG. 5: A number of Palm programs let you create and play back music. *Palm Tracker* from Emperor Studios (left) creates the illusion of multiple tracks; miniMusic's *BeatPad* is a pattern-based sequencer with an array of features.

you to set up rhythm patterns. You can also set up accents or loops and enable MIDI Clock in the Preferences window.

MiniMusic solved the timing issue by using a special internal clock that looks for the nearest system clock during playback. That seems to work well—*BeatPad* is a fun metronome. Included with *NotePad* in miniMusic's Mobile Software Pack, *BeatPad* also comes with Swivel Systems' SG20 sound module.

MobileSoft-Labs *M.Play* has an attractive 3-D interface that includes basic tape-deck controls for playing single-line MIDI files (see **Fig. 6**). The software lets you pick tunes from a library of prerecorded songs or create your own. Although *M.Play* is free, *M.PlayEdit* costs $10. With *M.PlayEdit,* you can create Palm-compatible files from MIDI files and transfer them from your Windows-based desktop computer to the Palm. A demo version is included with *M.Play*. I created a MIDI file of one of my fiddle tunes with ease.

Tom Zerucha's free *PlayMIDI* is available in two versions (see **Fig. 7**). With the older version, *PlayMIDI45A*, I could play the files that came with the program, but I couldn't figure out the instructions for converting regular MIDI files to files *PlayMIDI* could read. The color was also odd, because the program wasn't created for a color Palm.

The new version of *PlayMIDI* is part of a suite of programs called ZboxZ that includes compression utilities, fax software, and other small programs. Unfortunately, ZboxZ lacks a real installer,

so you must go through a rather arcane set of steps to install the software. In the new version, the files will not play without an external MIDI device, but you can actually hear the sounds as they were intended rather than be limited by the Palm's speaker. I still had problems creating files that would port properly to the Palm (probably because I use the relatively new *HotSync Manager* 4.0), and I had to manually rename some files before *PlayMIDI* would play them correctly.

If you use *PlayMIDI,* you will need plenty of memory: the program uses compressed files, and insufficient memory can cause a crash that requires a hard reset. I finally made the program work properly, and it should be able to play a variety of MIDI files as well as just music files. *PlayMIDI* is apparently the only program that lets you create multiline MIDI files on your desktop and port them to the Palm with an included utility applet.

If you're a fan of the theremin, you'll love *Theremini* from Pete Moss; it lets you create the effect of a theremin on your Palm handheld device. If you don't have a MIDI interface, use the Palm's internal speaker and make all sorts of weird and cool sounds by moving the stylus across the screen. The *x*-axis controls pitch; the *y*-axis controls volume. The left and right fields determine the frequency range (from 1 to 20,000 Hz); either field can be higher or lower than the other.

If you have a Palm MIDI adapter, you

can use the program to send out MIDI data through a MIDI interface. I used *Theremini* to play sounds directly from the Swivel Systems SG20 and ran the output into my computer, which opened up a new world of sounds.

## KEEPING SCORE

Among notation products for the Palm, Eric Cheng's simple *PocketSynth* (see **Fig. 8**) is probably the oldest in common use. You enter notes by tapping a note value on a palette (a Rest button is also provided) and then tapping a key on the four-octave keyboard. The notes play back and show up as text values on a line at the bottom of the page. A note is represented by a symbol such as "c02," which plays the note C in the bottom octave with a quarter-note value. Because the note information is just text, it can be entered or edited with a text editor such as Palm's *MemoPad*.

A slider sets the tempo, but the time and key signatures cannot be specified. You can export the notation information to Arkkra Enterprises' *Mup* desktop-music typesetting program using an intermediary Palm freeware program called *ToMup* from RMIT University.

## Music Hardware for Palm PDAs

| COMPANY | HARDWARE | WEB SITE | PRICE |
|---|---|---|---|
| **Handi-Q** | HandiClip HC2 | www.modularsound.com/handiq.htm | HandiClip $149; other cables $15 |
| **Handi-Q** | HandiClip HotSync-to-MIDI cable | www.modularsound.com/handiq.htm | $30 |
| **miniMusic** | Palm-to-MIDI interface | http://minimusic.com/midi.html | included in Mobile Software Pack |
| **Swivel Systems** | SG20 sound module | www.swivelsystems.com | $200, including miniMusic software |

# Music in the Palm of Your Hand

*ToMup* can combine as many as four melody lines and lets you enter time and key signatures, chords, lyrics, and other musical information. You can then export the files to *Mup* and print the music or export it as a MIDI file.

The two Palm software titles most like those common on regular computers are *NotePad* and TobelStudio's *TS Noter*. The programs let you store songs and work with one at a time, and they offer keyboard and staff entry. They also allow playback as the notes are entered or afterward through the Palm speaker or through MIDI to an external device.

Although less well known than *NotePad*, *TS Noter* has features that make it just as valuable as its competitor. It's an intuitive and straightforward program with a variety of useful features. In *TS Noter*'s Edit View (similar to Score mode in *NotePad*), you enter notes on a two-staff score by selecting a note value from a palette. The palette includes values from a whole note to a 64th note and also provides triplets, rests, flats, sharps, natural signs, and an Eraser tool. Once a note is entered, you can select one of the note values and tap on the note head to change it. You can drag the note up or down to change the pitch, and you can tie or slur notes together. Bar lines are entered in the same way: by tapping on an icon and then placing the bar line in the score. Although you can't specify a time signature, you can set the tempo in the metronome screen.

Additionally, the palette includes an arrow that allows you to insert numbered markers. You can later access those reference points with the Go To tool. Items in the File menu let you beam songs to another Palm and export files as alarms to be used in other applica-



FIG. 6: MobileSoft-Labs' *M.Play* offers a colorful set of 3-D tape-deck controls for playing MIDI files.

tions. (*Alarms* are audio or visual alerts that you can set to occur at specific times within other programs.)

The Edit menu provides the usual cut and paste functions and allows as many as ten undo levels. You can transpose all or part of a song up or down by as much as an octave. A toolbar across the top of the window offers quick access to the tempo setting (using the metronome); playback, MIDI, and recording preferences; Play and Stop buttons; the Score view; and the List view. The Piano view has the same basic menus and toolbar and permits real-time recording of a melody on a keyboard. You can scroll to view and play almost four octaves. In the MIDI Preferences screen, you can choose the Palm's speaker or a MIDI interface for output. If you select General MIDI (GM), the program provides a list of patch names.

You can send files to your desktop computer with the optional *PalMusic Desktop* application for Windows. Just
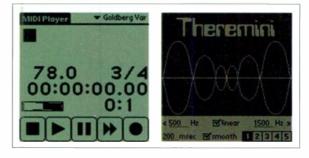


FIG. 7: Tom Zerucha's *PlayMIDI* (left) lets you create multiline MIDI files on a desktop computer and export them to the Palm. Pete Moss's *Theremini* produces the effect of a theremin on the Palm.

# Music in the Palm of Your Hand

open *PalMusic,* hit the Start button, put the Palm in its cradle, select Send to Desktop from the File menu, and then select Transmit. A dialog box asks where you want to save your MIDI file.

The Score View screen in *NotePad* looks similar to the one in *TS Noter,* with two staves for entering notation. With *NotePad,* however, you can enter as many as four voices, one a time; select them by tapping in the numbered box in the upper-right corner of the screen. You can even build chords within each voice. In the Voice Preferences screen, you can choose a voice's track, its MIDI channel, and an instrument (viewable by patch number, not GM name). Once back in the Score View screen, the Graffiti letter *V* toggles between "show four voices" and "show current voice." You

can also change voices by simply writing the number of the new voice in the Graffiti area.

Along the top of the screen, you can choose note and rest values ranging from a whole note to a 16th note. Enter notes by selecting a note value and placing it in the score. The note doesn't appear until you lift the stylus; however, you can hear where the note is as you move it around on the score. Unfortunately, once you lift the stylus, you can't use it to move a note, though a note can be moved by selecting it and using the Palm's Up-Down arrow button. You can also step-enter notes from a MIDI keyboard and shorten or lengthen a note in increments by drawing forward or backward strokes in the Graffiti area. Items in the Edit menu let you invert and reverse (retrograde) selected passages and convert notes to triplets. You can clear notes by selecting them and drawing an *X* in the Graffiti area.
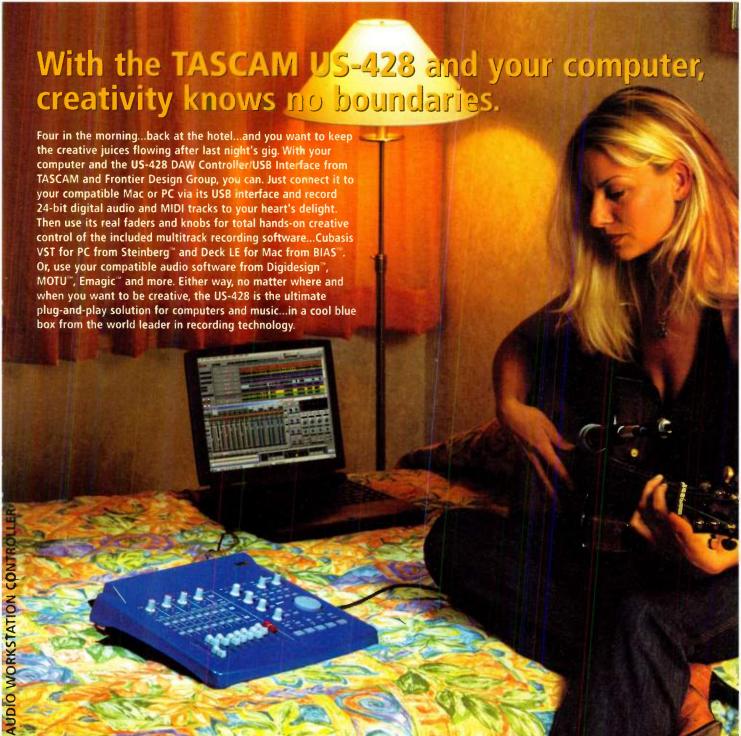
Using the Song Preference box, you can select the category, key, meter (3/4,



FIG. 8: Several programs let you turn the Palm into a handy musical sketch pad with basic notation capabilities. Shown here (clockwise from top left) are Eric Cheng's *PocketSynth,* TobelStudio's *TS Noter,* and miniMusic's *NotePad.*

4/4, 5/8, and 7/8), "play on entry," "show four voices," MIDI, and tempo. The time signature constrains how notes are entered, and placing a new note between existing notes might push them over to another measure to preserve the

## Music Software for Palm PDAs

| COMPANY/DEVELOPER | SOFTWARE | WEB SITE | PRICE |
|---|---|---|---|
| **Eric Cheng** | *PocketSynth* 1.24 (w/ *Metronome*) | www.echeng.com/Pilot/pocketsynth.html | $12.00 |
| **Wilson Cheng** | *MusicEar* 1.0 | www.musicear.com | $ 9.90 |
| **Gary Duke** | *Pocket Beat* 2.43 | www.aurete.com/pbeat | $ 8.00 |
| **Emperor Studios** | *Palm Tracker* 1.1 | www.emperor-studios.de | $15.00 |
| **Calvin Gaisford** | *Musician Tools* 1.1 | www.gaisford.com/mtools | $10.00 |
| **Handi-Q** | *HandiFX* 0.8 | www.modularsound.com/handiq.htm | $99.00 |
| **Handi-Q** | *HandiMix VCA* 0.1 | www.modularsound.com/handiq.htm | $99.00 |
| **Handi-Q** | *Ultra-Curve* 0.43 | www.modularsound.com/handiq.htm | $99.00 |
| **J. T. Solutions** | *MiniMidi* 3.0 | www.tsol.fsnet.co.uk/jts/minimidi/minimidi.html | $15.00 |
| **Mike McCollister** | *McChords* 2.0 | http://mikemccollister.com/palm | $10.00 |
| **miniMusic** | *BeatPad* 1.0 | www.minimusic.com/software.html | $29.95 |
| **miniMusic** | *NotePad* 1.1 | www.minimusic.com/software.html | $29.95 |
| **MobileSoft-Labs** | *M.Play* 1.0 | www.mobilesoft-labs.com | free |
| **MobileSoft-Labs** | *M.PlayEdit* | www.mobilesoft-labs.com | $10.00 |
| **Pete Moss** | *Theremini* 3.2.1 | http://users.ev1.net/~bantha/palm | $10.00 |
| **Palmetto Logic** | *Sound Advice* 2.01 | www.palmettologic.com | $ 7.95 |
| **Responsive Software** | *Responsive Metronome* 1.0.1 | www.responsivesoftware.com | $ 8.00 |
| **Geoff Smith** | *MidiMonitor* 1.0b | http://fargo.itp.tsoa.nyu.edu/~gsmith/ Pilot/PilotMidi.htm | $15.00 |
| **TobelStudio** | *TS Noter* 1.3 | www.tobelstudio.com | $28.95 |
| **TobelStudio** | *TS Tempo* 1.3 | www.tobelstudio.com | $10.00 |
| **Sam Trenholme** | *Ear Trainer* 1.0 | http://palm.samiam.org | free |
| **Tom Zerucha** | *PlayMIDI* (ZboxZ 0.25rc5) | www.execpc.com/~tz | free |

# Music in the Palm of Your Hand

meter. Play, Back, and Forward buttons are at the bottom of the screen along with icons to take you to the Piano and Grid views. The Piano view includes a condensed score at the top, so neither accidentals nor a written key signature appear. You can't enter notes directly on that condensed staff, but you can select notes from the palette and play them into the score from the piano keyboard. Switching to the Grid view displays a window much like a piano roll, with a keyboard along the left side as reference. You can still edit and play back the music in this view, and Grid view also lets you draw each voice in a different color.

One of the fun things about *NotePad* is that it includes 19 pieces of precomposed music that you can play back one line at a time using the Palm speaker, or you can play all the lines at once using an external MIDI device. *NotePad* is also part of the miniMusic Mobile Software Pack along with *BeatPad*, *Theremini*, and the Palm-to-MIDI interface. You save files to your desktop computer through a conduit added into the Palm Desktop software. *NotePad* also lets you beam songs to other Palm users.

Neither *TS Noter* nor *NotePad* imports MIDI files, though both export files that you can adapt with another program to replace Palm system alarms. But that's another hack.

## PACKIN' HARDWARE

MiniMusic's Mobile Software pack includes the Palm-to-MIDI interface (see **Fig. 9**). To set it up, connect a serial cable, cradle, or travel adapter to the serial port at the bottom of the Palm and connect the other end to the Palm-to-MIDI interface. A MIDI cable connects the interface to your computer or the MIDI In

port of a MIDI device. You then open a MIDI-compatible program on the Palm, make sure MIDI is turned on, and hit Play. I hooked up the Palm-to-MIDI interface to the MIDI In port on my PC and played through Voyetra's *MIDI Orchestrator Plus* with no problem.

The Swivel Systems SG20 is a small battery-powered tone module that clips onto the Palm's base at the serial port. As wide and deep as the Palm, the SG20 measures less than 2.5 inches in length. The SG20 has two output ports: one for headphones and one for an RJ11 cable. You can play MIDI applications and listen through the headphone jack, but that's just the beginning. When you plug the provided RJ11 cable in to its port, you can connect a small device that has MIDI In and Out ports, a MIDI-activity indicator light, and two RCA output jacks. In my tests, both units worked admirably.

The SG20 boasts a complete GM sound set and all the polyphony and features needed to make it fully GM1 compatible. Its onboard sounds and versatile design make it suitable for serious Palm-based composition. Other small battery-powered tone modules, such as the Yamaha MU15 and Roland PMA-5, attach to the Palm through the serial port, but unlike the SG20, they weren't created specifically for the Palm.

Handi-Q's HandiClip is an ambitious hardware and software system for remotely controlling high-end effects, mixers, and equalizers from the comfort of the Palm (see **Fig. 10**). According to Handi-Q's Web site, the system works with several devices, including



FIG. 9: Hardware devices include miniMusic's Palm-to-MIDI interface (left) and Swivel Systems' SG20 General MIDI sound module.

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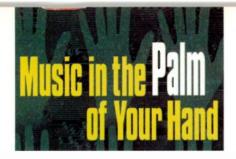After all, it is better to give than to receive.

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CITY                                                                STATE

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weeks for delivery. Name on sales receipt must match name on rebate coupon.                                          EM

the TC Electronic TC 1128, M2000, M3000, M5000, M-One, and D-Two; the Ashly Protea and VCM-88; the Behringer Ultra-Curve; the Yamaha O1V, O3D, SPX 990, and ProR3; the Lexicon PCM 70, PCM 80, PCM 90, and MPX 1; various MIDI synths and drum machines; and some intelligent lighting-control consoles.

The HandiClip attachment looks much like the SG20 (a standard shape for external Palm modems as well), except that it has a light that blinks red or green to show MIDI activity. It comes with an RJ11-to-KLR A3M adapter cable and an XLR-to-MIDI Y-cable. The RJ11 (telephone) cable plugs in to the port on the bottom of the HandiClip and attaches to the XLR-to-MIDI Y-cable. You can then plug the MIDI connectors in to the unit you wish to control. I plugged the setup into my PC, and it functioned the same as the Palm-to-MIDI interface and the SG20. The adapters let you plug the HandiClip in to longer RJ11 cables or in to a snake for controlling your units from a distance. The XLR cable can be used with the Protea using the RS232 or MIDI protocol.

Handi-Q has also just released its new HotSync-to-MIDI cable, which costs considerably less than the HandiClip. It offers a direct connection from the Palm serial port to a MIDI-controllable device through a MIDI Y-cable. Green and red LEDs on the MIDI connectors indicate data presence; simultaneous in and out flow is supported. Because the interface lacks batteries, it drains the Palm's batteries quicker than the HandiClip does.

According to the documentation, the HotSync cable performs successfully when connected through XLR cables on a snake cable run of 750 feet, but for anything longer than 600 feet, Handi-Q recommends the HandiClip. The HotSync cable also lacks the HandiClip's protection for the Palm from "improper cabling mishaps." In my tests, the interface worked just fine, but unlike most of the other cables, the ends are

FIG. 10: Handi-Q's HandiClip (top) allows you to control external hardware devices from the Palm. Handi-Q's less expensive HotSync-to-MIDI cable lets you connect the Palm to a MIDI-controllable device.

# Music in the Palm of Your Hand

female, so you need regular MIDI cables to connect it to your equipment.

## IN CONTROL

To use the Handi-Q products, choose the apparatus you wish to control and then download the appropriate Handi Systems Controller Software (see **Fig. 11**). Within the software, select a MIDI channel for each device that you want to control and adjust the parameters using faders. In the EQ software, you can set the faders in real time. You can also run in unsynchronized mode in case the EQ settings on the unit get changed while you are away from it. Then, hit the Synchronize button to send the information to the unit. In Handi-Q *Ultra-Curve,* the software's EQ curve is added to or subtracted from curves already in place on the EQ device. The documentation for the *HandiMix VCA* control software states that you can use the program with any equipment that has insert capability.

Because I use effects devices with my electric violin, I tried *HandiFX*. It lets you choose and name the mechanism you wish to control, set the MIDI channel, assign the controller number, and specify the minimum, maximum, and default settings. You then switch to the main screen to operate the faders attached to the selected device. You can

save the settings to a Palm Memo file for retrieval later and back them up to your desktop computer when the Palm is synced. I tried that with an MPX 500 from Lexicon, and it worked great. It won't send Program Change messages, but that capability is coming in a free update. Although the programs can be downloaded from the Handi-Q Web site, they only send out MIDI data after you enter the HandiKey (available after you pay for and register the product).

*MiniMidi,* from J. T. Solutions (see **Fig. 12**), simply sends messages to MIDI devices; it's more basic and less expensive than the Handi-Q products. *Mini-Midi*'s first screen is the Main/Keyboard page, which has a keyboard at the top to test the selected MIDI channel settings. A small horizontal slider lets you choose the MIDI instrument for the selected channels in the line of boxes below. The vertical faders just below send continuous controller messages to the controller, which you choose from a pull-down menu that includes three presets and four user settings. You can save settings on four pages. All the settings can then be sent at once, on the appropriate MIDI channels, to a number of MIDI-controllable devices. The program also lets you send different controller data on the same MIDI channel. You can set the controller
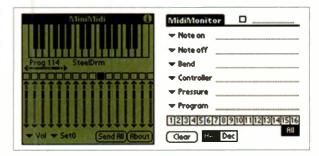


**FIG. 12: J. T. Solutions'** *MiniMidi* **lets you send messages to MIDI devices; Geoff Smith's** *MidiMonitor* **displays incoming MIDI data.**

number and the name for each field in the setup screen. As with the previous section, four banks of settings can be sent simultaneously. I tested *MiniMidi* on my PC, and it worked fine.

Geoff Smith's *MidiMonitor* lets you monitor incoming MIDI messages using a number of configurable fields. Data can be received in hex format, in decimal format, or as raw MIDI data. I tested *MidiMonitor* with the volume and wah pedals on a Line 6 Floorboard sending messages in to a Pod Pro and out the back as MIDI data. Although the program displayed continuous controller MIDI data, it did not seem to be consistently accurate. That was also the case with MIDI data from my PC. *MidiMonitor* may eventually be very cool, but at the moment, it's somewhat unreliable.

## COOL TOOLS

If you'd like to learn more about music-related products for the Palm OS platform, I recommend the following Web sites: the Handheld Music Homepage (www.crudites.org/soundventures/handheld-music) and the Composing Music on the Palm page (www.cs.mu.oz.au/~winikoff/palm/music). Most programs mentioned in this article are available as shareware at PalmGear (www.palmgear.com) or Handango (www.handango.com).

*Cat Taylor is an eclectic electric and acoustic violinist specializing in Celtic rock. Her current bands include Fiddler's Fancy, the Veil, and Avalon Rising. Special thanks to Larry the O for his technical assistance.*



**FIG. 11: Handi-Q's Handi Systems Controller Software provides a full complement of onscreen controls for several types of hardware devices.**

*Every year disguising presents from the members of my household becomes increasingly difficult. Often a simple shake of a wrapped package is all it takes to reveal the contents, which has forced me to repackage presents in oddly shaped boxes to keep them a secret as long as possible. The perfect accomplice in discreet gift giving is the stocking.*

*It may seem that a stocking can hold only relatively small, inexpensive items. However, to crafty gift givers, stockings offer a great place to hide an expensive gift from a wily recipient.*

*For this year's EM holiday gift guide, I decided to focus on small but practical items that anyone with a personal studio would enjoy. But unlike previous years, I have increased the price range to include as many of the latest gizmos as possible.*

*Let the wish lists begin!*

# A Studio in Your Stocking

1. Every studio needs a multitrack recorder. For a pocket-size personal digital studio, look no further than the Korg **ToneWorks PXR4** ($500). This miniature 4-track records directly to SmartMedia cards and offers 32 virtual tracks, a multi-effects processor, amp and mic modelers, and rhythm patterns. The editing features include track bouncing and time compression/expansion. The PXR4 also includes a built-in microphone and a ¼-inch input with a high- and low-impedance switch. Korg USA, Inc.; tel. (516) 333-9100; Web www.korg.com.

2. Soundman Microphones **OKM II K Classic Studio** ($325) is a pair of miniature omni mics you wear in your ears. The mics are light and fit comfortably in the ears, and the result you hear is a binaural stereo recording. Soundman offers seven models to cover a variety of recording situations, including rock, jazz, and classical music. Soundman Microphones/Independent Audio (distributor); tel. (207) 773-2424; e-mail info@independentaudio.com; Web www.independentaudio.com.
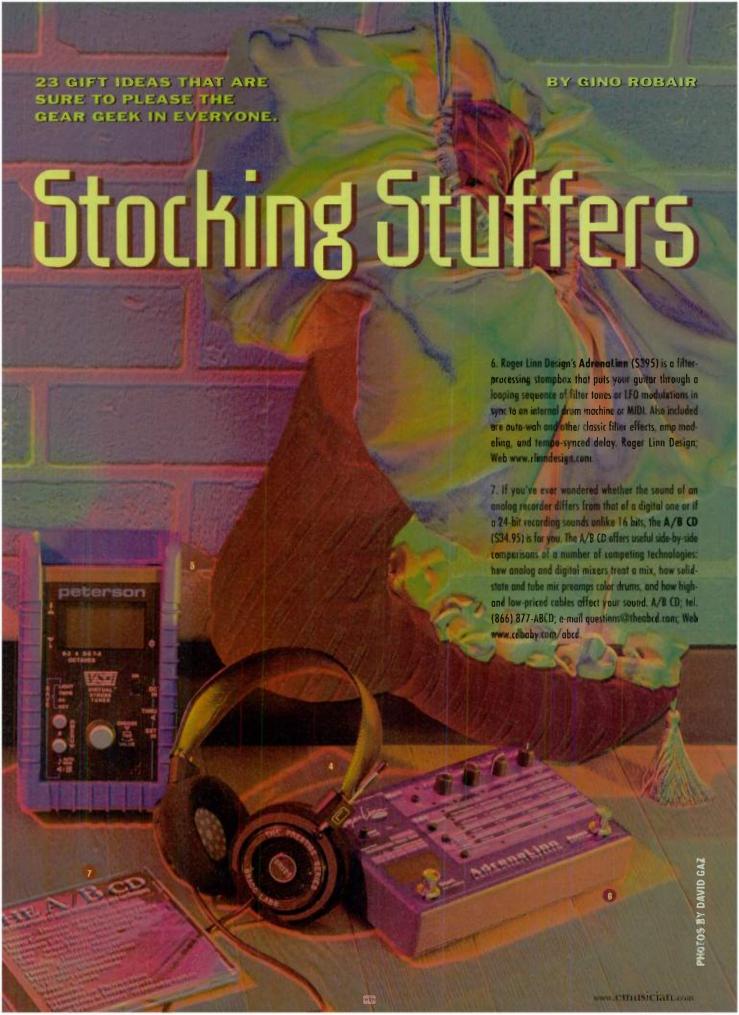
3. Whether you are practicing complex time signatures or laying down a quick click track, you can't top the Boss **Dr. Beat DB-88** ($195). The DB-88 is a programmable metronome that offers five divisions of the beat, a simple tuner, tap tempo, and two stereo headphone jacks. The unit runs on a 9V battery or a wall wart and can be attached to a cymbal stand. Roland Corporation U.S.; tel. (323) 890-3700; Web www.rolandus.com.

4. Every studio needs a pair of reference headphones. Grado's Prestige Series **SR125** headphones ($150) are the choice of many professionals. They offer excellent sound reproduction and a frequency range of 20 Hz to 20 kHz. Their lightweight design makes them comfortable to wear, especially during long sessions. Grado Labs; tel. (718) 435-5340; e-mail info@gradolabs.com; Web www.gradolabs.com.
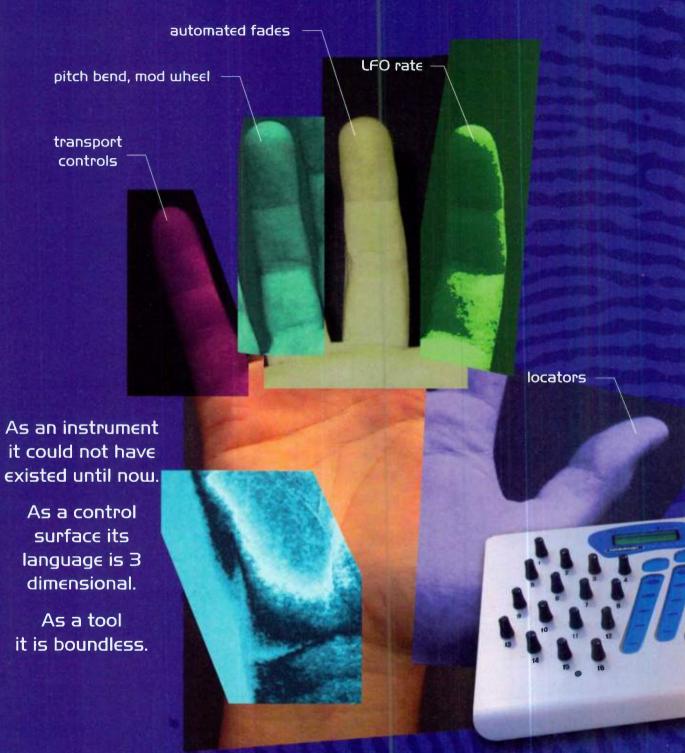
5. The Peterson **VS-1 Virtual Strobe Tuner** ($329) can tune everything in your studio. Accurate to 1/1,000 of a semitone, the VS-1 has a built-in mic, a ¼-inch input and output, nine octaves of reference tones, historical temperaments, and a reference A adjustable in 0.5 Hz increments. It runs on three AA batteries or a wall-wart power supply. Peterson Electro-Musical Products; tel. (708) 388-3311; e-mail email@petersontuners.com; Web www.petersontuners.com.

23 GIFT IDEAS THAT ARE
SURE TO PLEASE THE
GEAR GEEK IN EVERYONE.

BY GINO ROBAIR

# Stocking Stuffers

6. Roger Linn Design's **AdrenaLinn** (\$395) is a filter-processing stompbox that puts your guitar through a looping sequence of filter tones or LFO modulations in sync to an internal drum machine or MIDI. Also included are auto-wah and other classic filter effects, amp modeling, and tempo-synced delay. Roger Linn Design; Web www.rlinndesign.com.

7. If you've ever wondered whether the sound of an analog recorder differs from that of a digital one or if a 24-bit recording sounds unlike 16 bits, the **A/B CD** (\$34.95) is for you. The A/B CD offers useful side-by-side comparisons of a number of competing technologies: how analog and digital mixers treat a mix, how solid-state and tube mic preamps color drums, and how high- and low-priced cables affect your sound. A/B CD; tel. (866) 877-ABCD; e-mail questions@theabcd.com; Web www.cdbaby.com/abcd.

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pitch bend, mod wheel

LFO rate

transport controls

locators

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As a control surface its language is 3 dimensional.

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**Surface One** is a completely unique virtual control surface that uses a fiber optic touch-sensitive material as its primary interface.

The user-configurable trigger objects can represent virtual faders, buttons, and knobs, and can be linked to multiple messages. All objects can be assigned to any MIDI commands.

Surface One introduces no latency to your MIDI messages, and is compatible with any MIDI-compatible software programs or hardware.

# Santa's Little Controllers
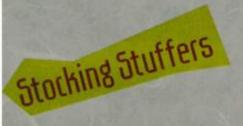
8. Whether you are tweaking the filters on a synth or adjusting parameters in a DAW, you will get better results using a real knob. Doepfer's **Pocket Dial** MIDI controller ($229) has 16 "endless" rotary controls and four banks for a total of 64 virtual knobs. For the desktop musician who has little open desk space, Pocket Dial's size makes it a winner. Doepfer Musikelektronik GMBH/ Enport (distributor); tel. (402) 398-0198; e-mail enport@ home.com; Web www.doepfer.com.

9. From C-Mexx comes **MIR** ($365), which stands for MIDI Interactive Remote. The space-age controller features a five-button touch pad, three control knobs, and a 2-by-16-character LCD for displaying the parameters you're working with. MIR fits easily in the palm of your hand or mounts conveniently on a stand. It is programmable and can import settings from other controllers. C-Mexx Software/X-Vision AudioUS (distributor); tel. (330) 747-3857; e-mail info@xvisionaudio.com; Web www.mircontrol.com.

10. Controlling virtual faders with a mouse is rarely satisfying. That's why Encore Electronics created the **Slidemate** MIDI control surface ($249). Slidemate features eight faders, four Group buttons, a Scene button (which also serves as a panic button), and MIDI delay. The device is fully programmable and can send continuous controller and System Exclusive messages. Encore Electronics; tel. (925) 229-8875; e-mail sales@ encoreelectronics.com; Web www.encoreelectronics.com.

# Holiday for Laptops

11. The Sound Devices **USBPre** (Mac/Win; $695) proves that great things do come in small packages. This 2-channel device includes a pair of balanced mic preamps with phantom power and a 15 dB pad; ¼-inch and RCA inputs; S/PDIF I/O; and a pair of headphone jacks. The USBPre uses 24-bit converters and has a frequency range of 10 Hz to 22 kHz. Sound Devices, LLC; tel. (608) 524-0625; e-mail info@sounddevices .com; Web www.sounddevices.com.

12. Building on the success of its low-latency Hammerfall PCI card, RME has created the **Hammerfall DSP Digiface** (Mac/Win; $650). This portable half-rack digital interface gives you 32 channels of MIDI I/O and 24 channels of 24-bit, 96 kHz digital-audio I/O. The connectors include front- and back-panel MIDI ports, three Lightpipe inputs and outputs, S/PDIF I/O, word-clock I/O, and an ADAT sync input. RME Intelligent Audio Solutions/X-Vision AudioUS (distributor); tel. (330) 747-3857; e-mail info@xvisionaudio.com; Web www.rme-audio.com.

13. No multichannel USB audio interface is smaller or sexier than Emagic's **emi 2/6** (Mac/Win; $499). This translucent blue interface provides two analog inputs, six analog outputs, and S/PDIF I/O in a case the size of a VHS cassette tape. A host of top-panel status LEDs keep you in the know during a session, and the emi 2/6 can be operated at 44.1 or 48 kHz at a resolution of 16 or 24 bits. It comes with ASIO drivers for Mac and PC. Emagic USA; tel. (530) 477-1051; e-mail emagic@ emagicusa.com; Web www.emagic.de.

# The beast in a titanium cage.

THE NEW *Fantom* WORKSTATION

76 KEYS · OVER 1000 SOUNDS · D-BEAM CONTROLLER · SRX-EXPANSION ·
FLEXIBLE LINEAR AND PATTERN RECORDING · REALTIME PHRASE SEQUENCING ·

14. Griffin Technology's **PowerMate** (Mac/Win; $45) is a deceptively simple USB device for controlling nearly any function. The polished aluminum knob has a smooth and solid feel when turned or pressed. It measures roughly one-and-a-half inches tall and two inches in diameter, so it can be placed just about anywhere. The skid-free base glows blue when your computer is on and pulses when your computer is sleeping. Griffin Technology, Inc.; tel. (615) 399-7000; e-mail sales@ griffintechnology.com; Web www.griffintechnology.com.
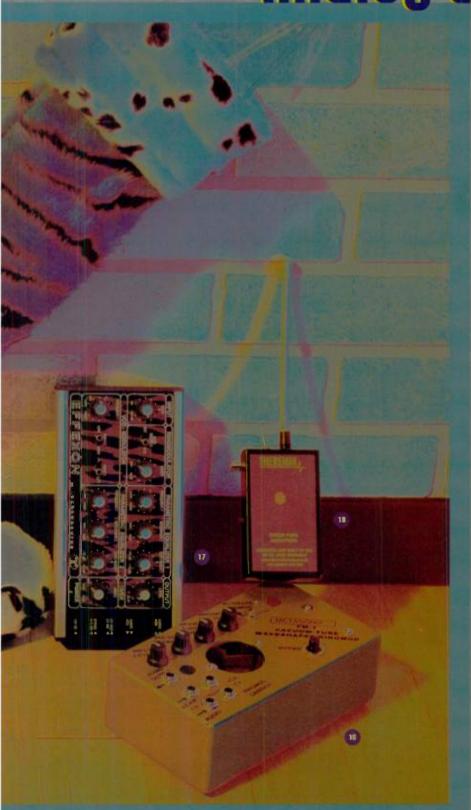
15. If you want to listen to ten hours of music from a player that fits in your hand, look no further than the Archos **Jukebox HD-MP3 Recorder Player** (Mac/Win; $349). It has a 6 GB hard drive, connects to your computer with a USB cable, runs on four rechargeable AA batteries, and comes with a carrying case, headphones, Mini Stereo/RCA adapter, and MusicMatch MP3-encoding software. Archos Technology Inc.; tel. (949) 453-1121; e-mail us-sales@archos.com; Web www.archos.com.

16. The **Computer Sound Morpher** (Win; $49) from Intel Play features a handheld recorder that can capture four minutes of sound with its built-in adjustable microphone. You can transfer the sounds to your computer using the supplied USB cable. The Computer Sound Morpher includes Windows 95/98–compatible software for editing and syncing your audio to animated onscreen images. You also get more than 90 sound effects to get you started. Intel Corporation; Web www.intelplay.com.

# Toys for All Ages

# Analog Christmas
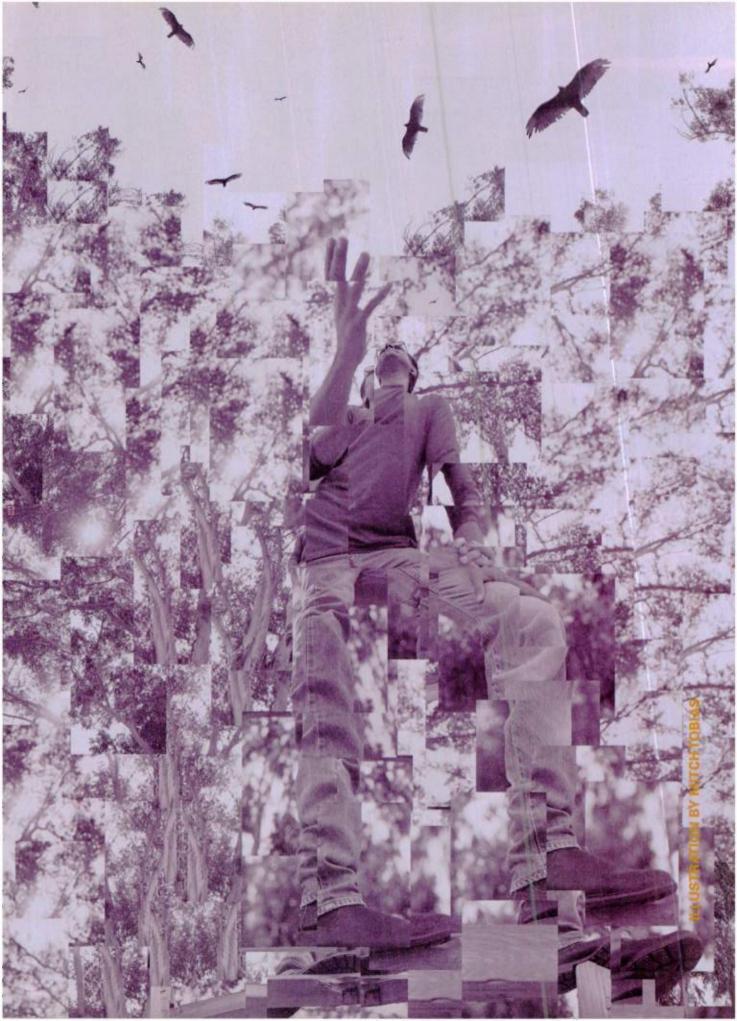
**17. Effexon** ($329), the latest in the Technosaurus line of Small Monsters, is a single-channel effects processor that combines two bands of parametric EQ with a voltage-controlled oscillator, ring modulator, and overdrive circuit. The overdrive circuit is normalized between the two EQs, so you can add severe squawk to almost any source material. Technosaurus/Drum Machine Museum (distributor); e-mail mickeyt@ drummachine.com; Web www.drummachine.com.
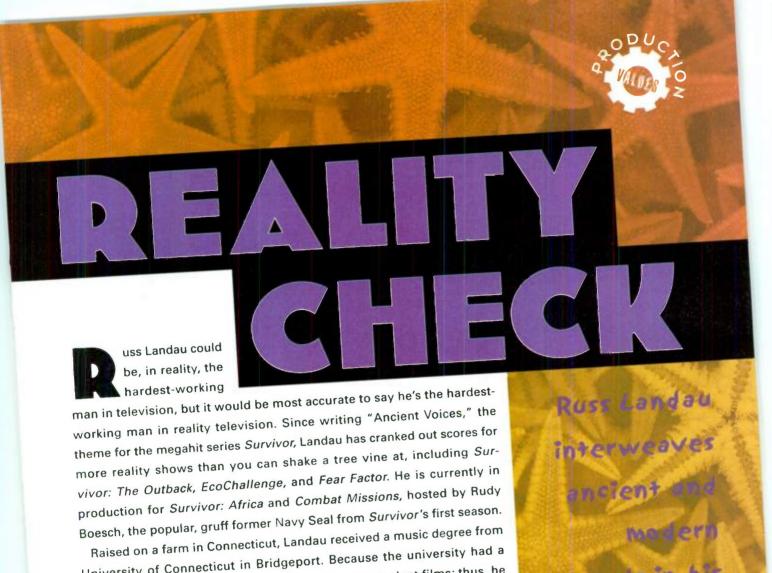
**18.** The Metasonix **TM-1** ($399) processor is unique for its size and for its functions. It combines a vacuum-tube ring modulator with the tube-based waveshaper from the TS-21 Hellfire Modulator, in a stompbox-size package. It's also designed to fit into the frame of a Synthesis Technology MOTM analog modular synth. Sounds processed through the TM-1 are just as distorted and unpredictable as you would expect. Metasonix; tel. (707) 263-5343; e-mail synth@metasonix.com; Web www .metasonix.com.

**19.** From the United Kingdom comes Rosedene Audio's appropriately named **Pocket Theremin** ($150). Although small, the battery-powered, space-controlled instrument includes a pitch antenna, a built-in speaker, an RCA output jack, and a sensitivity knob. Run it through a few effects devices—the TM-1 and Effexon will do just fine—for some screaming new sounds. Rosedene Audio; tel. 44-174-782-0536; e-mail rene@fullerton .demon.co.uk; Web www.fullerton.demon.co.uk.

# The Virtual Stocking

20. Whether you want to back up a hard drive or make a demo, every studio needs to burn CDs. Ahead Software's *Nero* 5.5 (Win; $69) supports 74-, 80-, and 90-minute media, has drag-and-drop functionality, and comes bundled with an MPEG-1 encoder for creating VCDs; *Cover/Designer*, a program for creating inserts and labels; and *Toolkit* for testing and modifying the speed of your CD-ROM drive. Ahead Software AG; tel. 49-724-891-1800; e-mail sales@ahead.de; Web www.nero.com or www.ahead.de.

22. With *Plasma* (Win; $69), Cakewalk offers musicians interested in loop-based composition and remixing a set of multitrack audio and MIDI tools taken from its flagship program, *Sonar*. Besides its beat- and pitch-matching features, *Plasma* accommodates DirectX Instruments software-synth plug-ins, such as *DreamStation* (bundled with the program), and includes *FXPad*, a plug-in that lets you use a joystick to control DirectX effects. Cakewalk; tel. (888) CAKEWALK or (617) 423-9004; e-mail sales@cakewalk.com; Web www.cakewalk.com.

21. MES *iSynth* (Mac; $79) is a fun and easy-to-use software synth that includes a step sequencer, a mixer (with mute automation), an arpeggiator, and effects. The sounds are generated by a pair of samplers, a bass synth, a mono synth, and a drum module. One-bar patterns can be chained together to create songs, and you can control *iSynth* in real time via MIDI using OMS. MES/Cycling '74 (distributor); tel. (415) 621-5743; e-mail info@cycling74.com; Web www.cycling74.com.

23. Magix *PlayR Jukebox XXL* (Win; $49.99) offers the budding remixer a chance to edit and combine audio and visual material in real time. You can set your music to a JPEG slide show or marry it to a live video feed. *PlayR Jukebox XXL* includes an audio editor, audio and video effects, and much more. A CD of 1,100 sounds is also included. Magix Entertainment Corp. USA; tel. (888) 866-2449 or (310) 656-0644; e-mail info@magix.net; Web www.magix.net. ⊛

# SO POWERFUL IT'S FRIGHTENING

PHOTO: STEVE SESKIS © 2001

## CRAIG (#5) > SLIPKNOT

Don't let that nice smile fool you. Craig is one tough customer. Producing multitrack projects, original loops, sonic textures, sequences and samples for his band Slipknot is a demanding task. And doing it while on the road can be brutal. He needs a digital multitrack studio that can take a beating. That's why Craig uses SONAR™, the professional's choice for music production on the PC.

SONAR is the only software available that combines digital audio and MIDI recording, audio looping, DXi software synths, and automatable DirectX audio effects into a seamless music production system. Designed by Cakewalk, the leaders in Windows audio technology for over ten years.

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# REALITY CHECK

**R**uss Landau could be, in reality, the hardest-working man in television, but it would be most accurate to say he's the hardest-working man in reality television. Since writing "Ancient Voices," the theme for the megahit series *Survivor*, Landau has cranked out scores for more reality shows than you can shake a tree vine at, including *Survivor: The Outback*, *EcoChallenge*, and *Fear Factor*. He is currently in production for *Survivor: Africa* and *Combat Missions*, hosted by Rudy Boesch, the popular, gruff former Navy Seal from *Survivor*'s first season.

Raised on a farm in Connecticut, Landau received a music degree from University of Connecticut in Bridgeport. Because the university had a strong film school, he spent a lot of time scoring student films; thus, he gained his first experience writing music for picture. After college Landau headed to New York City, where he became a session bassist as well as a member of the Folk Squad, a group of musicians that backed up artists such as Judy Collins and Mary Travers. He also became a longtime mainstay of the proto-new-age/world-music ensemble the Paul Winter Consort.

Landau migrated to Los Angeles in 1990 and picked up some work but found it tough going. When his father's health was failing, he decided to pack up his family and belongings to return to the East Coast. As fate would have it, no sooner was the moving van loaded than he received a phone call to try his hand scoring the Steven Spielberg–produced series *SeaQuest: DSV*. Landau received an Emmy Award nomination in 1996 for his work on that show, and he hasn't stopped since, progressing from session player to one of Los Angeles's first-call TV composers in the process.

For *Survivor*, Landau melds recordings of traditional instruments and themes with a mountain of samples, orchestral and otherwise, to achieve the ancient-futuristic blend that drives the show. He received an Emmy nomination this year for "Theme for *Survivor: The Australian Outback*." He shares the series-scoring duties tag-team style with composer David Vanacore.

*Russ Landau interweaves ancient and modern sounds in his scores for Survivor and other reality programs.*

**BY LARRY THE O**

# REALITY CHECK

In the studio of his bucolic Topanga Canyon home, the interview was highlighted by Landau's excited discussion of his then-upcoming trip to Kenya to do field recording and on-site composition for *Survivor*'s third season and was punctuated by appearances of his two sons. Clearly a happy musician, Landau has struggled, worked, and finally made it to a place where he can make his mark and live his dream.

**Why did you begin doing music for picture?**
Ever since I had the first opportunity to do music for visuals, I loved marrying music and picture. I love writing music, but it's much better to have it heard than have it just exist in a vacuum. I found that in writing for picture, even though music wasn't the primary focus, my music was heard: there was an audience, and I got to have some kind of response from people.

**How did you hook up with Paul Winter?**
Gordon Johnson, a friend of mine and an incredible musician, was Paul Win-



Most of Landau's television scoring work is done in the three studios attached to his house.

ter's bass player. He took another gig, so I subbed for him. He kept doing his gig, and I kept doing my gig, and it turned into a ten-year relationship where I went from bass player to bass player–writer–producer.

Gordon brought me into the band; then Paul Halley, Winter's keyboard player and "master composer," hired me. He was based at St. John's Cathedral in New York, and I started helping him with all his projects there. We became very close, and I produced his first album for Paul Winter's label. I coproduced [Winter cellist] Eugene Friesen's album, and then I started producing Paul Winter because I was good in the studio. It became very convenient for them because I played bass and manned the tape recorder.

What I'm doing now seems to be following in Winter's tradition. In Australia I went into a cave—a huge underground cathedral, if you will—recording on location in the homeland of the aboriginal musician David Hudson. It was inspiring to me in a similar way to what Paul did in the Grand Canyon and Lake Baikal.

**You must have been introduced to many ethnic and world influences.**
Yeah. We'd do a concert in Israel, and we'd jam with Israeli and Palestinian musicians. We did a concert in Brazil, and it was great to meet all those fantastic Brazilian musicians. I feel that music transcends boundaries, and the boundaries that are imposed on music are often demographic ones: a record label will ask, "Is it pop, dance, hip-hop, urban, rap?" It's commercial stuff. The most incredible thing for me is that with music for picture you don't get pigeonholed into any demographics; you can do whatever fits the scene.

**How did you get *Survivor*?**
I met Mark Burnett [one of *Survivor*'s executive producers] through our wives, who got into a fight over a parking space at the Montessori school and then became best friends, as those things usually go. Mark was producing *EcoChallenge* at the time. I really



Just home from school, Landau's oldest son, Ryan, gets an opportunity to play at the piano with Dad.

wanted to do his show, but that didn't happen then, because *EcoChallenge* was produced by another production company.

Then he got this show from England, *Survivor,* and he wanted to do his version of it in the United States. I started working with him on budgets a year before it ever got into production. As it got closer and closer, I started working on themes for it. Sixteen themes. Big orchestral themes. But they weren't really saying what the show was about. Eventually all were used, giving the show a big orchestral element.

Finally, we realized we needed to come up with a [common] musical language. Mark doesn't speak music in the same way I speak music. We decided to do a family vacation and drove on a ski trip to Mammoth [in California]. Our wives and kids all went in one car, and Mark and I went in another car and just listened to music and talked about music for six hours. We developed a way of speaking that made sense. After we came back from that weekend trip, I put "Ancient Voices" on his desk. Everybody flipped because it was the marriage of tribal with modern—*Lord of the Flies* meets MTV.

**One of the noticeable things from the first season was the blend of traditional percussion with synthesized textures. Do you make a conscious**

# REALITY CHECK

effect to do that, to juxtapose a modern context with the traditional context?

Sometimes. In the theme, for sure. I have electronic percussion going alongside traditional percussion; I have an ancient Russian folk song as the motor interwoven with a modern orchestral arrange-

ment. I just had the ancient Russian chant translated into Swahili and some lesser-known languages, and I'm weaving some little elements in and out [for *Survivor: Africa*]. For me it's tremendous. I love being able to explore and hear the Africans singing the chant. People

---

## RUSS NEVER SLEEPS

In his 15 years of professional playing and recording, Russ Landau has crossed some diverse musical terrain. Here are the highlights of his career in producing, scoring, and playing.

### Film Scores

*Senior Week* (Vestron, 1988)
*Voices from the Attic* (Siren, 1988)
*Cat and Mousse* (nominated Grand Prize, Cannes; HBO, 1989)
*Project: Tin Men* (movie cut; Orion, 1991)
*Bonnie and Clyde* (Fox Television, 1992)
*Other People's Secrets* (Rich Pix, 1992)
*Viva Terra Viva* (Modern Pictures International, 1992)
*Sea Scapes* (PowerSports Productions, 1994)
*Visions of the Orient* (Millennium International, 1995)
*The Rock* (additional music; Simpson, Bruckheimer/
  Hollywood Pictures, 1996)
*Telling You* (Division I/Cinetel Productions, 1997–98)
*Nowhere Land* (Prosperity Pictures, 1998)
*Love and Action in Chicago* (Prosperity Pictures, 1999)
*Totally Irresponsible* (Prosperity Pictures, 1999)
*Racehoss* (Breezeway Entertainment, 2000)

### Television Scores and Themes

*Fortune Dane* (pilot; ABC/Orion TV, 1987)
*Down Delaware Road* (pilot; NBC/Weintraub, 1989)
*Bodyguard* (pilot; NBC/Weintraub, 1990)
*Project: Tin Men* (pilot; ABC/Weintraub, 1990)
*Ren and Stimpy* (score; Games Productions/
  Nickelodeon, 1993)
*Carrier: Fortress at Sea* (Discovery Channel, 1994)
*SeaQuest: DSV* (12 episodes; Amblin/NBC, 1994–95)
*SeaQuest 2032* (Emmy Award–nominated score;
  Amblin/NBC, 1995–96)
*L.A. Firefighters* (20th Century Fox, 1996–97)
*Moloney* (Tristar/CBS, 1996–97)
*Baby Animal Tales* (Stone House/Discovery Channel, 1997)
*Four Corners* (score; Tristar/CBS, 1997–98)
*The Net* (Winkler/Tristar/USA, 1998–99)
*Combat Missions* (NBC, 2001)
*Fear Factor* (NBC, 2001)
*Survivor: The Australian Outback; Survivor: Africa*
  (CBS, 2001)
*Survivor U.K.* (British ITV, 2001)

### Jingles, Commercial Score, Logos

Kirin Beer (with Paul Winter, 1990–91)
Coors/Zima (O & M, 1993)
"PowerSports Theme" (logo; PowerSports Productions
  Syndicated, 1993)
Papazian Hirsch (logo; Papazian Hirsch, 1997)
Calgary Lane (logo; Calgary Lane, 1997)
Winkler (logo; Irwin Winkler, 1998)
Good Neighbor Pharmacy (Engle & Murphy, 1999–2000)
Make-a-Wish NASCAR (Make-a-Wish Foundation, 2000)
*Survivor* (logo; Burnett Productions, 2000)

### Record Production

In Action, *Missing* (Island/Tuxedo, 1986)
James Johnson, *Never Enough* (Island/Tuxedo, 1986)
Cheri Brandon, *Sleeping with the Enemy* (Geffen, 1987)
Ilene Kristen, *Give Away* (Island/Tuxedo, 1987)
Paul Halley, *Piano Song* (Living Music/AG, 1989)
Paul Winter Consort, *Wintersong* (Living Music/AG, 1989)
South Side Johnny, *Highway to Your Heart* (Disney, 1989)
Guire Webb, *New Frontier* (Proxima/BMG, 1989)
Cheri Brandon, *Too Late to Be Good* (Geffen, 1990)
Paul Winter Consort, *Earth: Voices of a Planet*
  (Living Music/AG, 1990)
Paul Winter Consort, *The Man Who Planted Trees*
  (Living Music/AG, 1990)
Paul Halley, *Angel on a Stone Wall* (Living Music/
  AG, 1991)
Rise, *Rise* (Proxima/BMG, 1991)
Lui Collins, *North of Mars* (Molly Gamblin, 1992)
Eugene Friesen, *New Friend* (Living Music/AG, 1992)
Guire Webb, *Leona* (Proxima/BMG, 1992)
Paul Winter, *Solstice* (Living Music/AG, 1992)
Ol' Paint, *Ol' Paint* (Fresh Jazz/LMG, 1993)
Paul Winter Consort, *Solstice Live* (Living Music/AG, 1993)
Paul Winter, *Prayer for the Wild Things* (Grammy Award
  winner; Living Music, 1994)
Noirin Ni Riain, *Celtic Soul* (Living Music, 1995)
*Ancient Voices* ("Survivor Theme" remixes; TVT
  Records, 2000)
*Dinosongs: Poems to Celebrate a T. Rex Named Sue*
  (Stone House/Scholastic, 2000)
*Survivor* (soundtrack; TVT Records, 2000)

# 5.1 REASONS ELLIOT SCHEINER PREFERS YAMAHA MSP10s.

"The MSP10s are the most accurate active monitors I've ever worked on."

*Elliot Scheiner, DVD Audio authority*

After 33 years as an engineer/producer, Elliot Scheiner has become the guru of DVD Audio. For his 5.1 surround sound mixes, Elliot uses Yamaha MSP10 monitor speakers exclusively. "They are perfectly powered speakers; you turn them up and they're still wonderful. My mixes sound great as they're going down and even better outside the studio."

*The MSP10 also comes in highly polished Maple Sunburst.*

*SW10 Subwoofer*

The MSP10's dual amps drive 120 watts to lows/mids and 60 watts to the highs. Their frequency response carries out well past 40kHz, perfect for 24-bit, 96kHz masters. Plus, in Elliot's surround environment, "probably the most important aspect: the balances relate very well." Which, of course, makes them ideal for stereo mixes as well. Once again, Yamaha gives you more for less.

## ⊛ YAMAHA

CREATING 'KANDO' TOGETHER

# REALITY CHECK

often think it is an African chant and it isn't; it was Russian, but now it is African!

**So you get the continuity yet at the same time ...**
. . . a change in elements, and it's a major element to change.

*Survivor* is really fun because I get to do African music when I'm in Africa; I can do Australian-influenced music when I'm in Australia. So it's a palette with a framework. Obviously, I'm not doing traditional music; I'm doing my impressions from listening to the music, and then I take it back and use elements and try to do something that works well to picture.

**Talk a little about how you work with David Vanacore on *Survivor*.**
David and I are basically two independent composers that are working together on the series. There are two executive producers on the show: Mark Burnett, who created it, and Craig Piligian. Mark brought me in on it, and Craig asked me if I didn't mind sharing, seeing that there were 40 minutes of music to do on the show each week. He had a very talented composer that he wanted me to work with, and after I heard David's stuff, I thought he was great. So they put us together, and we've been having fun.

**Do you guys coordinate at all?**
Yeah, on some things. We spot the shows together and decide what [cues] we're [each] taking. We flip a coin sometimes.

**What about on an aesthetic level, coordinating the sorts of orchestration and ideas you're going to use?**
When we spot, we're both coming from a similar place in terms of our synth and orchestral scoring, but we bring different things to the table as far as that goes. We both like long-line melody; we both write cinematically.

David came up with the cool tribal council music that's used all the time, and I wrote the theme, so we each bring an element to the show. In some ways, with two of us scoring, there's a little friendly competition between us to see who can write the next cool thing.

**Did you go on-site at all for the first series?**
No. I had no interest in going to Borneo. Last year, for Australia, it was not in the budget for the composers to go, but David and I decided that we both wanted to go. It worked out so well that [sending the composers on-site] is in the budget now. This year *Survivor* is flying me to Kenya.

It's great because I go on-site for five days in the middle of [production] and score a bunch of stuff for the editors. As soon as they start shooting, they're editing, and they like to edit to music, so I dump all the stuff that I'm doing right onto the Avid [video-editing systems].

**When you go down there, what kind of rig are you going to take?**
I'll be recording for five days in the field on a PowerBook G3/500 with a Sound Devices USB pre as well as an HHB PortaDisc recorder. We'll be documenting it on digital video, which will also be recording sound as a backup. Then, I'll have five days on the *Survivor* site where I'll be composing. I'll be using my PowerBook running Emagic *Logic [Audio]* with *Logic Software Instruments* and *EXS24* as well as [BitHeadz] *Unity DS-1*.

I have a little Yamaha keyboard, and Yamaha is also providing me with one of its new Compass Series guitars to bring down so that I can jam with the tribes. Hopefully it won't get wrecked—stampeded by a herd of water buffalo or something.

**What are you going to be recording in the field?**
I'm going to 12 different remote villages and recording their ancient folk music. I have a guide, Anindo Marshall, a former pop singer from Kenya who's well loved and recognized, and she'll be my liaison. She's taking me to all the tribes and helping me with translations.

**Are you going to get singing and instrumental music?**

Mostly I'm interested in the old tribal traditional stuff that I can incorporate into some of the more modern things that I'm doing. The truth is that Los Angeles has a fairly strong population of Kenyan musicians. I've had some of them over here already working on arrangements and recording; I can do that here. I want to get the raw materials there. I want to find the real ancient, rough stuff: the



Singers from the Samburu tribe in Umoja village listen to takes recorded on the HHB PortaDisc recorder. On the left is Daniel Kip Tanui, Landau's Masai guide.
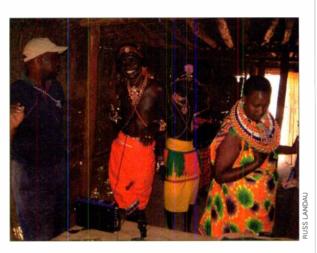
kids' group that's singing the traditional tribal song from 1,000 years ago. I want to learn from that and use it in what I'm writing. I want to put the viewer in the bush. It's as much of a learning experience for me as it is for the viewer.

**The show is currently done in stereo. Nothing in surround?**

No, just in stereo. Most people have mono TVs, a lot of people have stereo TVs, and a growing number of people have surround systems on their televisions, but [the score] better sound good on the lowest common denominator. When I do a show in surround, a lot of times when it's played back on stereo TV, the music disappears or one element of the music pokes through in a way that's very strange. I don't feel

that the standardization of down mixing is really there yet.

We did mix a version of the theme in surround, and it was fun because we got to do all kinds of cool effects. As far as the music goes, left-center-right is where we would be anyway in a surround system. We would put other things like ambience in the surround speakers, but not the music. If I'm in a movie theater, I don't really want to hear the music coming from behind me; I prefer the center spectrum. That way the music becomes part of the picture.

I have two ears; I'm kind of a stereo person.

**From what you say, you're able to work in a traditional post fashion in which you can spot the footage, so you're not writing blind.**

I am writing a lot of music blind to give them music to cut to. A lot of music is earmarked for certain kinds of scenes, and the editors get to choose from a library's worth of material.

**Picture editors, not music editors? That seems unusual.**

We have a great team of editors, and it's really the only way a show like this could work. That's why the music pops so well: the picture is cut to it, so a lot of it plays like a music video.



In Kenya the Kangemi tribe greets Landau with native songs.

# REALITY CHECK

I'm really enjoying it. Sometimes I'm surprised in a painful way, but if I don't agree with something, I can change it. I have that option.

Then, when the picture's locked, we have a spotting session and decide what stuff has to be replaced or where they couldn't find perfect cue in the library. If I see something I can do better, I'll say, "I can score that with a lot more emotion." So it's a combination of traditional scoring and letting the editors have their way with a library of original music that's been composed for the show.

*Survivor: Africa* is taking up a lot of time right now because of all the conceptualization of what I am doing with the theme and the languages and how to incorporate the Masai stomps and heavy breaths with the Swahili chanting.

## SURVIVAL GEAR

At press time, Russ Landau was expanding the number of his studios at his home from two to three, adding, subtracting, and moving gear between the three rooms in the process. As a result, he wasn't able to give actual quantities on some of his gear, but the bulk of his working rig is given below. He adds that his arsenal has a few constants: "Always, a Sadowsky bass, a Taylor guitar, and an old Fender Strat [are] by my side. But I really love the new Yamaha Compass edition guitar, which traveled with me throughout Africa."

**Computers**
Macintosh G4s (2); 9600 (1)
Emagic AMT8s
Emagic Unitor8s
MOTU MTP2 MIDI interface
MOTU MTPAV MIDI interface

**General Recording Setup**
AKG C 414 EB through Avalon VT-737
Daking 52270H preamps
Grace Model 201 preamps

**Hardware**
Akai S5000s (a couple)
Digidesign Pro Tools/24 Mixplus (room A)
Digidesign Pro Tools/24 (room B)
Digidesign Pro Tools Digi 001 (room C)
E-mu E4 samplers (lots)
Kurzweil K2500s; K2600s (a mess of them)
Kurzweil K2600 as controller
Mackie Digital 8-bus console
Mackie LM-3204 submixers (plenty)
Tascam *GigaStudios* (a few)
Yamaha ProMix mixers

**Mixdown**
Alesis M20
Alesis Masterlink
Panasonic SV3700
Tascam DA-88

**Monitors**
Dynaudio Acoustics BM6A
Tannoy 6.5II
Yamaha NS10s

**Remote Setup**
AKG C 1000 mics
AKG C 480 B mics
Digi 001 chassis
Earthworks M30BX mics
HHB PortaDisc MiniDisc recorder
Mac PowerBook G3/500
Magma chassis
Sound Devices USBPre 1.5 preamp

**Software**
BitHeadz *Unity DS-1*
Emagic *EXS24*
Emagic *Logic Audio*
Digidesign *Pro Tools*
MOTU *Digital Performer*

### How do you research that?

I have my Kenyan experts; they come over, and we talk. I'll throw a rehearsal together with some Kenyan singers, and we'll just experiment with the microphones on. I'm messing with stuff. I've worked with a lot of Africans throughout my career as a bass player, so I've been dying to sink my teeth into this stuff. The groove is forever.

The last episode of *Survivor [Outback]* had a live portion of the show. This was maybe the coolest thing I've done in my career. I scored the last 15 minutes of *Survivor* into the next hour of the Bryant Gumbel portion when he interviews all of the cast members. They determined the winner with a live vote on live TV all over the world, and I scored it with an on-camera 13-piece ensemble, conducted and played.

We were at CBS in Television City [Los Angeles], and it was a live gig in front of 70 million people. It was really nerve-wracking and scary but incredibly exciting. That went over so well that I'm already booked for the next live *Survivor* gig.

### Talk a little bit about orchestration and tools. What is the mix of stuff that you do—synths versus sampled instruments versus instruments performed live?

For the live orchestra performance, I had a sample of a sawtooth loop playing, and that was my click track to keep everybody together as well as a major element in the theme. So I am holding that down in one hand, playing a sort of a soft pad in my left hand, and then everyone is playing along with it. So I'm totally mixing in the synthetic elements, even live.

### What about day to day?

I bring people in once a week for a recording session, but most of the framework of the cues is done between my E IVs, *GigaStudio*, Kurzweils, and *EXS24* using a wealth of samples that I've bought. The Spectrasonics stuff is incredible, but with samples you have to be the first guy to use them; otherwise, you've heard them before, so I try to create a lot of my own samples and percussion loops.

# REALITY CHECK

Members of the Masai tribe in Tino village pose with Landau after recording the new intro to the *Survivor* theme.

It's fun working with loops; it's a good way to get a little motor going instead of a click track playing. Often I'll change the loop after I'm done writing to it. I don't always love writing to loops. I do it when I need what I call a "tasking cue" that needs a groove motor running through it. Otherwise, I just love putting my hands down and writing what comes to mind out of the blue.

**You're using a lot of different samplers: you have the E-mus, an Akai, a Kurzweil, and *Giga-Sampler, EXS24,* and *Unity*. What makes you choose one rather than another?**
I have them set up to do different things. My brass and a good section of my strings are living on the E-mus. A lot of my synthetics and choirs are on the *GigaSampler. GigaSampler* is great because I can dump everything in there; it's the bin for every sample I have. It can't play all of them at the same time. It does a pretty good job on a fair number of voices, but it's always there when I need it. It's great because I can have this palette of 1,000 sounds available to me all of the time without having to go stick in a sample CD, and I can audition them really fast.

*EXS24* is sort of like that; I'm just getting involved in that. For my portable stuff, I need *EXS24* and *Unity* because I can't take all my racks.

The more software synths and samplers you try to cram through a single output of an audio device, the more it all starts to sound kind of mushy like an organ. So in the studio, I like to have my separate outputs.

**Do you sequence everything and mix down from that, or do you print the samplers and synths back to hard disk and mix from those tracks?**
I mix from the MIDI. I don't see any reason to dump all the audio on. The only reason to print all the synths as audio would be if I wanted to take it to another studio and mix it on an SSL.

**Getting those exact sounds might also be easier.**
Yeah, but there's something kind of cool about the random nature of what could happen doing it again. It's a score, so it's a very liquid medium. It's never really done until you say it is. It's kind of the same thing with a song: it's subjective.

**The difference being broadcast dates, which don't move.**
Yeah, deadlines. It's done. We love that. It limits you. And really, does the music change that much? Is it going to have that much of a different effect on you if I brought up the EQ on the hi-hat another decibel? I'll hear it a little bit, but is it going to change the impact of the cue?

I'll sit and mix for hours, and sometimes the first demo rough mix is the one you go with, because it has the most life before you kind of EQ the life out of it.

**Do you have any parting words about your work?**
Everything I do is for one reason only: that I hope to get to meet Poe. *[Laughs.]*

When you're doing music for hire, the hardest thing is to find your own identity. You have to be flexible and not be attached to the individual things that you write. Eventually, you're going to end up working for people that hire you because they love your sensibility and love that little special thing that you bring to their projects. But you can't jealously guard your notes. You're writing for hire, and the grace in what you do will come out in subtle ways. It's a collaborative effort, and your producers will be your collaborators. You have to help them to be successful. The hardest thing to give up is that jealous protection. It's just music. [Landau's son] Max asked me, "Dad, why is music so much fun?" It doesn't all have to be so important; it should somehow be fun. It is the best job in the world if you can do it and have fun with it.

**Larry the O** *is a musician, producer, engineer, and sound designer in the San Francisco Bay Area. He has been a contributor to* **EM** *since 1986.*

Landau's portable recording setup works hard while on location in Kenya.

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Take a good look at the MRS-1044. Its power and simplicity bring the spontaneity back to digital recording. While you may not be able to see many of the dozens of features that make it so incredible, its greatest asset is the way they all fit together. It's the Zoom of desktop recorders.

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**Personal effects:** Process and sweeten the mix with legendary Zoom 24-bit digital effects, including EQ, Compression, Reverb, Chorus and even VAMS guitar amp modeling. There's also dedicated EQ on every channel.

**It's well connected:** The MRS-1044's open design provides enough ins and outs to satisfy any session. Phantom powered XLR balanced, 1/4 inch unbalanced inputs, stereo RCA analog outs, S/PDIF Optical out, MIDI and a slot for add-in SCSI and USB! No kidding.

**Zoom audio quality:** The MRS-1044 gives you Zoom's legendary audio quality with remarkably smooth, clean 24-bit A/D and D/A converters and 44.1 kHz bit resolution.

**More questions?** Obviously there's more than we're telling you here but we think we've already got your attention. With the MRS-1044 all you'll need is inspiration. Good luck.

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## The Zoom MRS-1044 MultiTrak Recording Studio

# Musical Protocols

## Can't we all just get along?

By Brian Smithers
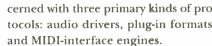
When shopping for new hardware and software, desktop musicians must wade through a veritable alphabet soup of audio and MIDI protocols. Consumers are constantly confronted with a confusing array of abbreviations, from ASIO to WDM, representing technologies that may or may not be compatible. With so many protocols from which to choose, it's ironic that sometimes the word *standard* is used to describe them.

For the most part, a protocol is a clearly established, standardized method of handling communication. In a diplomatic context, protocols prevent people from offending folks with big weapons and bad attitudes. In a music-technology context, protocols provide for efficient and reliable handling of audio and MIDI data between hardware and software or between a host program and plug-ins.

Both Macs and PCs have such protocols built into their operating systems, but historically they have been geared toward general-purpose multimedia and gaming rather than critical audio production. The need to achieve consistent professional-level real-time performance with high-resolution multichannel audio is the driving force behind the development of third-party protocols.
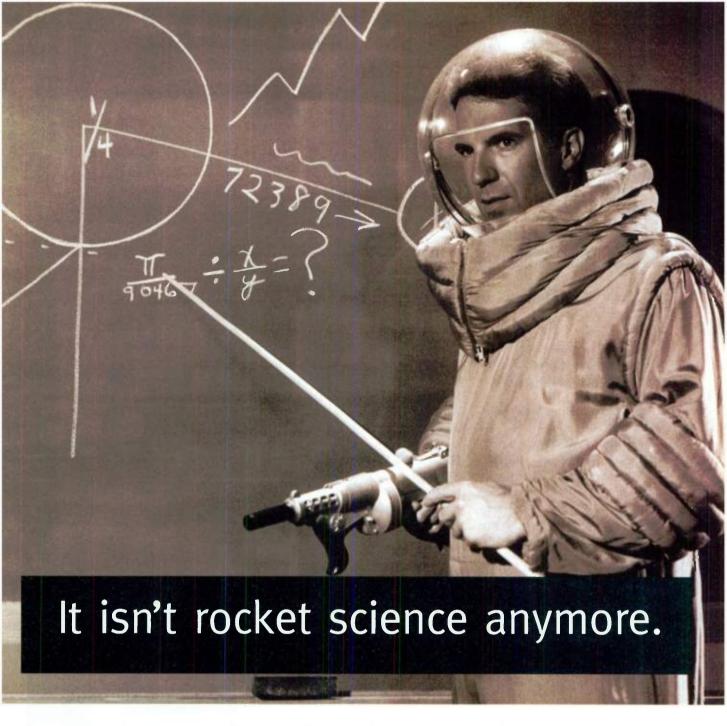
Desktop musicians must be concerned with three primary kinds of protocols: audio drivers, plug-in formats, and MIDI-interface engines.

### ASIO LIKE IT

Audio drivers are the bits of code that make hardware devices available to programs. No matter how different two audio interfaces may be at the hardware

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**FIG. 1: MOTU's** *Digital Performer* **lets you easily set the sample rate and buffer size when using the ASIO driver.**

level, they need to present their functions to the software in a uniform way, and the audio driver enables them to do that.

Probably the best-known effort to improve the state of computer audio drivers is Steinberg's Audio Stream Input/Output (ASIO) specification, which applies to Macs and PCs. In essence, it bypasses Apple's Sound Manager and the Windows MME layer and communicates directly with hardware to provide multichannel, high-resolution capabilities. A hardware manufacturer that writes ASIO drivers can expect any ASIO-compliant program to be able to communicate with the company's hardware, and software developers can expect the same.

In addition to making extra features available, ASIO improves on the performance of OS-level audio protocols. In particular, it succeeds in reducing audio-processing delays (latency) that have traditionally been a relatively low priority for the designers of operating systems. With Windows MME drivers, for example, it could take one-half to three-quarters of a second for audio to pass from an input to an output. ASIO brings latencies as low as 6 to 8 ms with current drivers. It accomplishes that in part by giving users control over their audio-buffer settings (see the sidebar "Loving Latency Lost"). The faster your system, the less read-ahead buffering it will need to keep up with the flow of audio, and the lower the latency will be as a result (see **Fig. 1**).

In another approach to reducing latency, many hardware manufacturers build direct input monitoring into their interfaces, allowing incoming audio to go immediately to an output without passing through the computer. That lets a guitarist hear what he or she is playing in real time along with the prerecorded tracks, for example, but it doesn't let you process the guitar with any software effects.

ASIO 2.0 addresses that problem by allowing you to bypass the hardware's direct monitoring and pass the input through the software. You can then process the signal with reverb and other plug-ins and monitor the audio with a manageable 6 to 10 ms delay. Version 2.0 also adds sample-accurate positioning for ADAT Optical transfers and provides for multiclient functionality, letting audio devices be shared among applications.

## NICE AND EASI

Although ASIO successfully delivered performance that was not possible with the default Mac or Windows audio drivers, Steinberg wasn't the only company working to make things better. Emagic developed its own solution, called *EASI* (Enhanced Audio Streaming Interface). EASI was announced after ASIO 1.0 was released, and it anticipated some of ASIO 2.0's improvements, which included sample-accurate synchronization. EASI was the audio driver protocol of *Logic Audio* starting with version 4.0, and Emagic opened it to developers as a potential standard.

Ultimately, ASIO 2.0 grabbed the spotlight, and EASI now enjoys only limited support. Some hardware manufacturers continue to provide EASI drivers, but in a bow to the market's realities, all *Logic Audio* versions also support the ASIO standard.

## MODEL DRIVER

Meanwhile, Cakewalk, unconcerned with cross-platform issues for its Windows-only *Pro Audio 9*, was convinced that the shortest path to superior performance was through the operating system itself. Instead of jumping on the ASIO band-

wagon, Cakewalk worked to convince Microsoft of the importance of pro-level audio support. As a result, Cakewalk's support for input monitoring and other features lagged behind its competitors. However, with the release of *Sonar*, the successor to *Pro Audio* 9, Cakewalk's efforts have finally paid off.

*Sonar* is the first application to take full advantage of the new Windows Driver Model (WDM), the core audio driver for Windows 98 SE/ME and Windows 2000/XP. WDM *kernel streaming* represents a huge leap forward for audio performance on a PC, providing a much more efficient and flexible link between hardware and software. WDM audio drivers finally solve, at the OS level, the problems for which ASIO and EASI had previously been the only solutions. For example, when I switched from the old MME drivers to WDM drivers on my Pentium II/266 MHz laptop system, my latency when playing a virtual instrument dropped from more than half a second to an imperceptible amount. The performance of WDM drivers is said to be especially good under Windows 2000.

Microsoft isn't alone in finally seeing the light. Apple has implemented a new set of audio application programming interfaces (APIs) called CoreAudio in



**FIG. 2:** *Cubase VST's* **Audio System Setup dialog box holds many settings, such as buffer size, that give users control of ASIO and VST performance. The Automatic Plug-In Delay Compensation checkbox (lower right) tells** *Cubase VST* **to automatically shift audio tracks earlier by the amount of time that plug-in processing would otherwise delay the output.**

Mac OS X, and its stated intention is to make third-party protocols (such as ASIO and EASI) unnecessary. Apple's list of objectives for CoreAudio includes goodies such as multichannel audio and application-determined latency, and an independent test shows that CoreAudio is indeed capable of impressively low latency.

Does that mean ASIO is in the position of answering a question nobody's asking? CoreAudio's impact is not yet clear, but many major hardware manufacturers are or soon will be offering WDM drivers, and other developers of Windows audio software are working on WDM compatibility.

Still, ASIO is familiar ground for many developers and consumers, and because a lot of companies make products for both Macs and PCs, it might make sense for them to keep ASIO around to facilitate cross-platform development. My crystal ball shows hardware manufac-

turers turning out three drivers for every audio interface—ASIO/Windows, ASIO/Mac, and WDM—at least until some consensus about WDM and CoreAudio is reached. (The same crystal ball misses the Lotto numbers week after week, so who knows?)

A couple of other terms you'll often hear in the same breath with ASIO, EASI, and WDM are MAS (MOTU Audio System) and DAE (Digidesign Audio Engine). They are horses of a different color, though: technically, they're hard-disk recording engines, not I/O protocols. As such, they function at a different level, communicating with audio hardware through the drivers. MAS, for example, can work with ASIO or Sound Manager.

### PLUGGED IN

Plug-ins operate at yet another level by assimilating into the host application. Plug-ins are everywhere these days, even popping up in music-notation programs. I'll focus on host-based real-time audio plug-in formats that are open to third-party developers, because that's where much of the action is. Of the major real-time native plug-in formats, two, Steinberg's Virtual Studio Technology (VST) and Microsoft's DirectX, are supported by multiple host programs (see the sidebar "Who's Plugged In?"). The other biggie is MAS (yes, that name does double duty). Although it's exclusive to Mark of the Unicorn's (MOTU's) *Digital Performer* and *AudioDesk*, it is supported by a significant roster of big-name third-party developers.

Although some people use the term *VST* to refer to *Cubase VST*, it actually applies specifically to Steinberg's cross-platform, real-time native plug-in format. VST defines the ways in which plug-in developers integrate their designs with VST-compatible host programs.

Like other native plug-in formats, VST provides a 32-bit floating-point architecture with automation and MIDI control. Those features let you record changes to plug-in settings and also sync effects to the tempo of your sequence. *Cubase VST*'s Automatic Plug-In Delay Compensation (see Fig. 2) can shift tracks slightly ahead to offset the

FIG. 3: Cakewalk's *Sonar* is the first program to support DirectX-based virtual instruments. DX Instruments (and VST Instruments) allow easier setup and better performance by integrating software synthesizers, such as Audio Simulation's *DreamStation DXi*, into the host program.

small amount of time required for plug-ins to process audio.

VST version 2.0 introduced support for *virtual instruments*, sometimes informally referred to as VSTi. In the past, software synthesizers ran as separate programs, and you had to redirect MIDI and audio between applications using so-called virtual cables such as the Mac's IAC Bus or *Hubi's Loopback* on the PC. By integrating software synths into the host application, VST 2.0 not only simplified the setup but also reduced the strain on system resources. It's not unreasonable to suggest that VSTi added momentum to the software-synthesizer revolution by making the technology so easy to use.

## THE DIRECT APPROACH

Microsoft's DirectX 8 is the other factor in the WDM and *Sonar* revolution. DirectX (technically DirectShow, part of the DirectX family) has provided basic real-time plug-in support for Windows audio programs for years, but now its feature set has been enhanced significantly.

Prior to version 8, DirectX just wasn't in the same league as VST or MAS. Although its 32-bit architecture allowed developers to write competitive-quality effects, DirectX's lack of support for automation and virtual instruments made it a less interesting format than the other two. That has changed, and DirectX 8 supports automation with subsample timing resolution.

Working closely with Microsoft, Cakewalk developed a way to incorporate virtual instruments into the DirectX plug-in environment. Cakewalk's DirectX Instruments (DXi) format was launched in early 2001 with support from several

software-synthesizer manufacturers, and the instrument list continues to grow (see **Fig. 3**). Like VSTi, DXi supports SysEx and provides for communication of patch names from the synth to the host program.

There has been a perception in the industry that DirectX plug-ins run less efficiently than their VST counterparts, and some circumstantial evidence has supported that impression. According to plug-in developers with whom I spoke, though, that is not the fault of the DirectX specification—VST and DirectX offer identical latency specifications in most cases.

With the major plug-in formats sporting such attractive features, the only potential downside for users would be having to choose just one. Fortunately, the major VST hosts support DirectX plug-ins in their Windows versions, and there are programs designed to make VST plug-ins operate under DirectX or MAS. That is accomplished by *wrapping* a plug-in in a thin layer of code that makes a VST plug-in appear to be a DirectX or MAS plug-in to a DirectX- or MAS-compatible host. One such program is *VST-DX Adapter* from FXpan-sion Audio. *Digital Performer* devotees can use Audio Ease's *VST Wrapper* to add VST plug-ins to their palettes.

In fact, plug-in developers may write for one format and then simply add a wrapper layer to make their plug-in function in the other format rather than completely rewrite the plug-in code. The wrapper layers can be implemented efficiently enough that performance degradation is negligible.

## BATTLING MIDI INTERFACES

It seems as though everybody is coming out with new MIDI-interface protocols these days. The good news is that they're typically cross-platform solutions; the bad news is that they're all proprietary.

Emagic led the way with AMT (Active MIDI Transmission), first implemented in its Unitor MIDI interface. In an effort to improve MIDI timing, Emagic devised a way to buffer MIDI messages in the interface before they are sent. That allows data at different ports to be sent simultaneously, instead of sequentially, for tighter timing.

Emagic claims accuracy of better than 1 ms when the Unitor is used with *Logic*

*Audio.* That is the sword's other edge. Although such precise computer-to-interface timing is great, it requires an interface capable of buffering the MIDI data and a compatible host program that can presend the data to the interface. That's why AMT and its rivals are proprietary systems. Use the Unitor with a different program, and AMT no longer works for you.

A conceptually similar timing scheme joins MOTU's *Digital Performer* with its line of USB MIDI interfaces. A hardware-based MIDI-transmission engine called MIDI Time Stamping (MTS) leverages the increased bandwidth of USB (over serial) to deliver what MOTU claims to be ⅓ ms timing between the computer and the interface. The latency for recording MIDI data is said to be even lower: only ⅓₂ ms. Like AMT, MTS makes use of data buffering within the interface. But MTS time-stamps every data event, regardless of which cable it's going to, so MTS affects the accuracy of every data event, not just data events distributed across multiple cables. *Digital Performer*'s MIDI resolution has been upgraded to take advantage of that precision, so now you can

## LOVING LATENCY LOST

Achieving the lowest latency your system can support without degrading audio performance requires balancing three factors: the Disk Block Buffer size, the number of buffers, and the amount of memory that is allocated for each channel. Fundamentally, the smaller the buffer size, the lower the latency. Keep in mind, however, that the smaller the buffer size, the closer to the ragged edge of disaster your system is running. If you hear audio degradation (clicking or dropouts) you should increase the buffer size or number of buffers. That requires a corresponding increase in the amount of memory that is allocated per channel, which can decrease the maximum number of channels your system will support (see **Fig. A**).

Steinberg's *Cubase VST* manual has detailed instructions for adjusting

**FIG. A: This dialog box in MOTU's *Digital Performer* lets you set the optimum buffer size and number of audio tracks to deliver the best performance for your system. The bottom of the dialog box indicates the required RAM for the current configuration.**

those settings and encourages experimentation to find the optimum settings for your system and needs. Some users deliberately set their buffers a bit too low when recording VSTi parts, enduring some sonic compromises in favor of more responsive timing. During mixing, they set their buffers higher, ensuring flawless audio performance despite somewhat slower response.

Your system's best performance will probably be achieved by using your audio interface's dedicated ASIO drivers (if it has them) rather than the ASIO Multimedia or ASIO DirectX Full-Duplex drivers. For information about audio cards that offer ASIO drivers, check out the web site http://service .steinberg.net/testbase.nsf or www .kvr-vst.com/asio.php.

adjust the resolution over a wide range and select microresolutions with as many as four decimal places.

Not to be outdone, Steinberg jumped into the fray with a similar technology called Linear Time Base (LTB). LTB is automatically enabled when you use Steinberg's Midex8 interface with the latest revision of *Cubase VST/32* 5.0. Like its counterparts, LTB makes use of time stamping and buffering to increase the accuracy of MIDI timing.

## STANDARD DREAMS

If, like me, you dream of genuine universal standards (of which MIDI itself may be the best example) this part of the story doesn't have a fairy-tale ending—at least not yet. The three proprietary MIDI timing schemes aren't likely to merge, but Apple's Mac OS X includes another little gem called the CoreMIDI API, which, like CoreAudio, is intended to make third-party standards obsolete. Its performance objectives include latency lower than 1 ms and very low jitter (timing variations). Windows reportedly also has such an infrastructure. If hardware and software manufacturers adopt these standards, everyone will benefit.

Now that the major audio and MIDI developers have raised the bar and, in doing so, have finally made Apple and Microsoft take notice, professional-level support for MIDI and audio is widely available through operating-system protocols and through various independent standards.
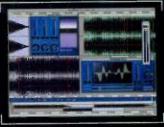
**Brian Smithers** *is the associate course director of MIDI at Full Sail Real World Education in Winter Park, Florida.*

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

# Microphonic Machinations

## Microphones are the door to the world of acoustic music.

By Scott Wilkinson

*During the past 15 years, many fundamental music-technology concepts have been explained in "Square One" (originally titled "From the Top"). In 1997 **EM** technical editor Scott Wilkinson combined many of those columns into a comprehensive primer titled* Anatomy of a Home Studio: How Everything Really Works, from Microphones to MIDI, *published by EMBooks, an imprint of Artistpro.com (www.artistpro.com).*

*Our readership has continued to grow, and new readers shouldn't be left behind. Rather than try to reinvent the wheel, we've decided to periodically reprint excerpts from the book in the form of "Square One Classics." These articles clarify the essential, unchanging concepts that make it possible to be an electronic musician.*

A microphone's job is simple: it converts an acoustic sound into an electrical signal that corresponds to the original waveform as closely as possible. That signal then can be processed, mixed with other audio signals, and recorded.

Acoustic sounds occur when something vibrates at a frequency between 20 Hz and 20 kHz in air or another medium. That creates small regions of high and low pressure around the vibrating source. As the molecules in those regions move in response to the changing pressure, they jostle nearby molecules, and that causes the regions of high and low pressure to expand outward.

Once the regions of changing pressure reach a microphone, they impinge on a flexible *diaphragm* within the mic, causing it to vibrate in response. That physical vibration is then converted into an electrical signal, which is sent to a mixer, a signal processor, or another device. The difference between types of mics is the specific manner in which the conversion is performed.

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## PHYSICAL ATTRIBUTES

Before I discuss the various mic types, here are some concepts that are common to all. As mentioned previously, all microphones include a diaphragm, which is usually mounted in something called a *capsule*. The capsule is mounted in an outer case along with any support electronics. Some cases can be handheld or standmounted for stage use, whereas others must be mounted on a stand. Stand-mounted mics sometimes include a shockmount, which isolates the mic from unwanted vibrations in the stand.

In most cases, the capsule is located behind a screen of some sort that lets the acoustic sound enter while protecting the diaphragm from physical damage. That screen often includes a layer of foam to reduce wind noise and vocal pops, though an external pop screen is usually more effective in the latter application.

Once the acoustic sound has been converted to an electrical signal, it is conveyed to another device along a cable. Some inexpensive mics include a permanent cable that terminates in a 2-conductor, ¼-inch or ⅛-inch phone plug. The cable includes one central conductor that carries the audio signal surrounded by another conductor (called the *shield*) that connects to ground. However, that type of cable is susceptible to induced hum and other environmental noise. As a result, most professional and semipro mics use a 3-conductor XLR connector at the end of a balanced cable.
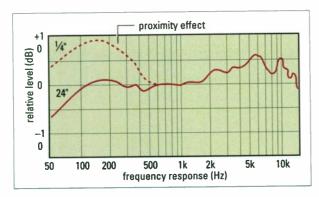
FIG. 2: Notice that the pickup patterns—(a) cardioid, (b) supercardioid, (c) figure-8—are slightly different at different frequencies. In these examples, the left half shows the curves for 125 and 500 Hz; the right half shows the curves for 1 and 4 kHz.
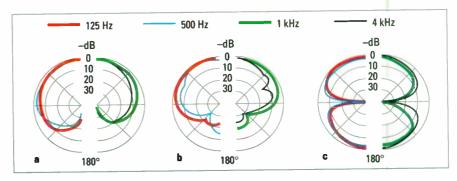
## FREQUENCY RESPONSE

The frequency range within which a microphone accurately translates the sound-pressure level (SPL) of acoustic sounds into electrical signal levels is called its *frequency response*, which is measured in decibels (dB) over a range of frequencies. But what does "accurately" mean? For a given SPL, the output-signal level typically varies by no more than ±3 dB from its nominal level. That is normally depicted in a graph of frequency versus output level (see **Fig. 1**). A mic with a flat frequency response generates the same audio signal level for a sound of any frequency within the specified range at a given SPL.

However, most mics don't exhibit a flat frequency response, partly because making such a mic is expensive and partly because a frequency response that's uneven can be of some benefit. For example, many vocal mics boost the upper frequencies; that is the *presence peak* shown in **Fig. 1**, and it helps improve the intelligibility of words. However, a presence peak can exaggerate a shrill upper vocal range.

At the low end, the frequency response of a vocal mic often falls off below 100 Hz. Because the human voice can't produce frequencies that low, there is no reason to make a mic that reproduces

them accurately. Instrument mics generally fall off below 50 Hz. However, the low-end response of many mics can be greatly enhanced by moving the sound source close to the mic. That bass boost is the *proximity effect* in **Fig. 1**, and it helps radio announcers achieve their characteristically deep sound. However, moving too close to the mic increases breath noises and vocal pops.

## PICKUP PATTERNS

All mics exhibit a *pickup pattern*, which determines how the mic responds to sounds at different frequencies coming from different directions. An *omnidirectional* mic responds more or less equally to sounds coming from any direction. That pickup pattern is particularly well suited for ambient mics, which are used to pick up the sound of the room in which an acoustic source is radiating.

In many cases, omnidirectional mics are not used in live performance because they pick up sounds from all directions, which can lead to feedback. However, omni mics are generally less susceptible to wind and breath noise, and they tend to have a relatively flat frequency response with no pronounced peaks, which can actually help to avoid feedback. Omni mics also tend to have excellent low-frequency response, and they do not exhibit the proximity effect.

If a mic does not respond equally to sounds from any direction, it is called a *directional* mic. There are several types of directional mics, most of which respond best to sounds coming from directly in front of the mic's capsule. (The main exception is the *middle-side*,

FIG. 1: A microphone's frequency response is depicted in a graph of frequency versus relative output level. If the sound source is close to most directional mics, the proximity effect boosts the low end. Also notice the presence peak between 2 and 10 kHz.

or M-S mic, which contains two capsules. M-S mics pick up sounds from both sides as well as the front.) Sounds that strike the mic at its most sensitive spot are *on-axis;* sounds from any other direction are *off-axis*. Directional mics are prone to the proximity effect, and their frequency response is normally less flat than in omni designs.

The pickup pattern of any microphone can be depicted in a *polar graph* (see Fig. 2). In this type of graph, the mic's axis is defined as 0 degrees (usually located at the top of the graph), and the outer circle defines a flat frequency response. The smaller, inner circles represent a drop in frequency response. The curve within the graph indicates how the mic responds to sounds from different directions. An omnidirectional mic's polar pattern

▼

## The most popular type of directional mic is the cardioid.

forms a circle. Keep in mind that although polar patterns are conventionally graphed in two dimensions from a bird's-eye view, the mic's actual pickup pattern is three-dimensional.

The most popular type of directional mic is the *cardioid* (see Fig. 2a). Its polar pattern resembles an inverted heart—hence the name. The mic is most responsive to on-axis sounds, whereas off-axis sounds are attenuated; sounds from 180 degrees off-axis are almost completely rejected. In addition, notice that the polar pattern changes slightly at different frequencies. The extra curves give a rough idea of a mic's frequency response and pickup pattern in one graph.

A *supercardioid* mic is often used in live performance because it rejects more sound from the sides than a cardioid design (see Fig. 2b). However, it does have some response to sounds coming from 180 degrees, as indicated by the small rear pickup lobe. Another
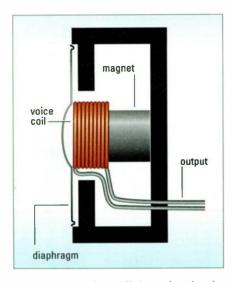
**FIG. 3: In a moving-coil dynamic mic, the diaphragm is attached to a coil of wire that vibrates within a magnetic field. That induces an electrical signal in the wire that corresponds to the incoming acoustic waveform.**

variation of that design, called the *hypercardioid,* is even more directional.

Some mics exhibit a *bidirectional* or *figure-8* pickup pattern (see **Fig. 2c**), so-called for obvious reasons. Those mics are most sensitive to sounds from the front and rear, rejecting sounds from the sides. That works well for miking two sources (such as two toms in a drum kit or two singers facing each other) with one mic.

## DYNAMIC RANGE AND MORE

Not only must mics reproduce different frequencies coming from different directions but they must also contend with sounds at different levels. If the sound reaching a mic is above a certain amplitude, the signal from the mic becomes distorted, and the diaphragm might even be damaged. This upper amplitude limit is called the *dynamic range,* and it's measured in dB SPL. The dynamic range of most mics is typically between 100 and 120 dB SPL; some go to 130 dB SPL or higher.

At a given source level, different microphones produce an audio signal at different levels. The relationship between the input level and the output level is called the *sensitivity* of the mic; the higher the output level at a given input level, the more sensitive the mic.

Sensitivity is usually measured with a 1 kHz tone at one or two levels: 74 dB SPL (the level of the average speaking voice at a distance of 3 feet) and 94 dB SPL (which corresponds to a loud speaking voice at a distance of 1 foot). The output level is expressed in dBV (decibels referenced to 1V root mean square) or dBm (decibels referenced to 1 milliwatt). (See "Square One: Decibels Demystified, Part 2" in the August 2001 issue for more about those decibel types.) For example, a mic's sensitivity might be specified as an output level of –47 dBV at 94 dB SPL. Many mics also include a pad switch, which lowers the overall output level by 10 or 20 dB. That is useful if the sound source is particularly loud.

Microphones also exhibit *impedance,* another important electrical characteristic. That is the microphone's resistance to the flow of electrical current, which changes as a function of frequency (see "Square One: The Shocking Truth" in the June 2001 issue). Most professional and semi-pro mics are low impedance (typically between 150Ω and 250Ω but sometimes as low as 50Ω; also called *low-Z*), whereas most inexpensive mics are high impedance (usually above 20 kΩ; also called *high-Z*).

Low-Z mics are less susceptible to extraneous electrostatic noise in the cable, such as that caused by fluorescent lights or motors, but they are more likely to pick up hum from electromagnetic interference, such as that from AC power lines. Because they operate with relatively high current levels and use balanced cables, low-impedance mics can drive cables that are hundreds of feet long. High-Z mics, especially those using unbalanced lines, are limited to cable lengths of no more than about 20 feet.

Matching a mic's impedance with the input to which it is connected is important. To connect one type of mic to the other type of input, you must typically use a *matching transformer*. In a pinch, you can connect a low-Z mic directly to a high-Z input, but you will lose too much level if you connect a high-Z mic to a low-Z input.

## DYNAMIC MICS

There are many types of microphones that use different methods to convert an acoustic sound into an electrical signal. Of those types, three are most common today: *dynamic, condenser,* and *boundary.*

In most dynamic microphones, the diaphragm is attached to a coil of wire, called the *voice coil,* which is free to move near a permanent magnet (see **Fig. 3**). As the diaphragm vibrates in response to an acoustic sound, the voice coil moves back and forth within the magnetic field. That induces an oscillating electric current in the wire, which corresponds to the original sound's waveform. The design is also known as a *moving-coil* mic. Common moving-coil dynamic mics include those in the AKG Tri-Power series; the Audix OM-3xb, OM-5, and OM-7; the beyerdynamic M 88 and TG-X series; the Sennheiser MD 421; and the ever-popular Shure SM57 and SM58.

Moving-coil dynamic mics are often housed in a handheld case. They are also quite rugged and able to withstand rough treatment and high SPL levels. As a result, they are great for miking drums, electric-guitar cabinets, and vocals. Most dynamic mics use a cardioid, supercardioid, or hypercardioid pickup pattern to reject onstage ambient noise. Good dynamic mics have an excellent frequency response, but the diaphragm–voice coil assembly is relatively heavy, so those mics are somewhat less sensitive to fast transients.

A variation of the moving-coil design is called the *ribbon* mic. Instead of a diaphragm and voice coil, that type of mic uses a thin ribbon of metal suspended in a permanent magnetic field. As the ribbon vibrates in response to an acoustic sound, an electric current is generated in the metal.

Although they're not made much these days, ribbon mics have excellent transient response, and they are famous for their warm sound. However, they are extremely fragile and delicate; you can destroy the ribbon by coughing into the mic. Their output level also is generally lower than moving-coil designs.

## CONDENSER MICS

In a condenser mic, the diaphragm is a very thin sheet of Mylar coated with gold or another conductive material and suspended over a parallel conductive surface called the *backplate* (see **Fig. 4**). That forms an electrical capacitor, which is sometimes called a condenser—hence the name. A static voltage of 9 to 48 VDC is applied across that capacitor. As the diaphragm vibrates in response to incoming acoustic sounds, the voltage varies slightly as the diaphragm moves closer and farther away from the backplate, generating a tiny signal. That signal must be amplified with an internal preamp before it is sent to the mic's output. Popular condenser mics include the AKG C 414, the Audio-Technica 4033a/SM and 4050/CMS, the beyerdynamic MC 834, and the Neumann TLM 193.

The static voltage across the condenser element and the power required to operate the internal preamp are typically supplied by a battery or by the input of the device (such as a preamp) to which the mic is connected. The voltage from the device's input is called *phantom power,* because it is sent along the same cable that carries the signal from the mic to the input; there is no separate power cable. That does not interfere with the audio signal, because the phantom power is a fixed voltage, whereas the audio signal changes over time.

A variation of the approach is called an *electret condenser.* In that type of mic,
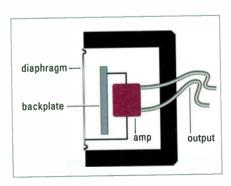
**FIG. 4: In a condenser mic, the diaphragm is suspended over a parallel backplate, forming a capacitor. As the diaphragm vibrates, the voltage across the capacitor varies, producing an electrical signal, which must be amplified by an internal preamp.**

the diaphragm or backplate is made of a material (such as Teflon) that retains a permanent electric charge, which removes the need for an external power supply. (However, the internal preamp still needs power, which is normally supplied by a battery.) In most cases, the backplate retains the charge, so the diaphragm can be made of Mylar instead of Teflon, which is less sensitive to acoustic vibrations. That design is called a *back-electret condenser.*

Condenser mics can exhibit different pickup patterns. Many condensers offer several switchable patterns, which is handy. Condenser mics have excellent transient response because the mass of the diaphragm is very low. They are most often used in the studio for vocals, acoustic instruments, and just about everything else. However, they are more delicate than dynamics, so they are not used as much in live performance.

## BOUNDARY MICS

Another variation of the condenser approach is the *boundary* microphone. In that design, a small electret capsule is mounted in the center of a flat metal plate, and that forms a sonic boundary—hence the name. Incoming sound waves are reflected from the plate and reach the capsule at virtually the same instant as the direct sound waves. That reinforces the acoustic signal at the diaphragm.

Crown's version of the boundary mic, called a Pressure Zone Microphone (PZM), has become the most common form of boundary mic in the United States (see **Fig. 5**). Unlike other boundary designs, in which the capsule faces away from the plate, a PZM capsule is suspended above the plate by a small distance and faces downward, toward the plate. The gap between the plate and capsule is called the *pressure zone,* where the direct and reflected sound

**FIG. 5: The Sound Grabber II is one of Crown's PZMs. The electret capsule is suspended over the plate, facing downward.**

waves meet and add coherently to stimulate the diaphragm.

In most boundary mics, the electret capsule is omnidirectional. However, the plate effectively blocks any sound coming from the side opposite the capsule, so the practical pickup pattern is hemispherical. These mics are often mounted on a large board or wall for picking up room ambience. They also work well on instruments with a large sound-radiation pattern, such as vibes or woodwinds. A few boundary mics have a cardioid pattern, in which case the capsule axis is parallel to the plate rather than perpendicular. These mics are often used on stage floors for actors.

Unless you record and perform instrumental music on synthesizers exclusively, you need at least one mic in your toolbox. However, mics can be expensive. Invest in at least one high-quality condenser for recording purposes and one or more dynamics for recording and live performance.

Each mic has its own unique sound, so experiment with different models until you find the right one for each instrument or voice. Once you find the right mics for your applications, the world of acoustic sound awaits your pleasure.

*The studio in which **Scott Wilkinson** records the voice-over narration for United Airlines' Classical Collection audio channel uses a hypercardioid shotgun mic aimed right at his mouth.*

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**▶ [====] start playing.**

# REVIEWS

## NATIVE INSTRUMENTS

### SPEKTRAL DELAY 1.0 (MAC/WIN)

*More delays than you'll find at the airport.*

By Peter Freeman

Native Instruments has already made significant inroads in the desktop-music market with *Reaktor* and other products. Now the company has produced yet another remarkable piece of software. *Spektral Delay* 1.0, which runs as a standalone application and a VST plug-in, takes the concept of frequency-dependent delays and turns it into a powerful sound-processing environment.

Like all of Native Instruments' products, *Spektral Delay* runs without dedicated audio hardware. Its flexibility and low cost come with a price, though: you need a serious computer to take full advantage of the program.

I tested the standalone application and the VST plug-in on a Power Computing clone with a 400 MHz G3 processor card with 272 MB of RAM. With the program looping a sound file and the modulation features engaged, the response of the program's controls was sluggish and unwieldy, bordering on unusable. (Keep in mind that the memory architecture on an upgrade card is slower than that on a new computer.) But the actual sound output remained excellent. On a G4/450 MHz, the program's performance was noticeably better, though occasionally the mouse response was still frustratingly slow.
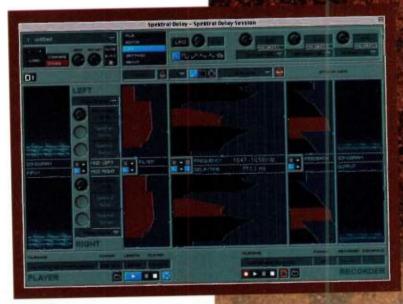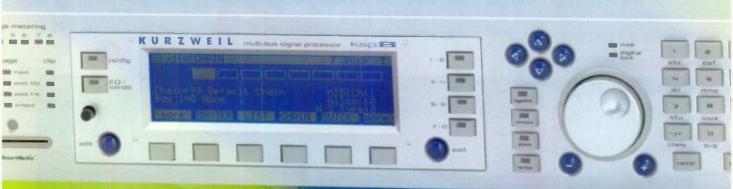
FIG. 1: *Spektral Delay*'s interface is a single clearly organized screen. The Matrix Editors (center) allow you to manipulate the three main parameter controls (Filter, Frequency, and Feedback).

www.emusician.com

**FIG. 2:** The Editor screen has several functions for manipulating the Matrix Editor controls. Among them are quantization and other mathematical functions, as well as an LFO with variable wave shapes.

Fortunately, you can tweak *Spektral Delay*'s settings in a number of places to optimize performance on your hardware. As is usual with plug-ins, the faster the computer, the happier you'll be with the program's performance.

*Spektral Delay* supports standard MME, DirectX, and ASIO drivers under Windows or Sound Manager and ASIO on the Mac. (Real Time AudioSuite support is under development, according to the manufacturer.) The program ships with a nicely printed softcover manual in four languages (English, German, French, and Spanish). In general, it communicates most of what's needed to use the program effectively. However, I wish more were included about the program's MIDI control features, specifically more about using *Spektral Delay* with Open Music System (OMS) on the Mac via the IAC bus.
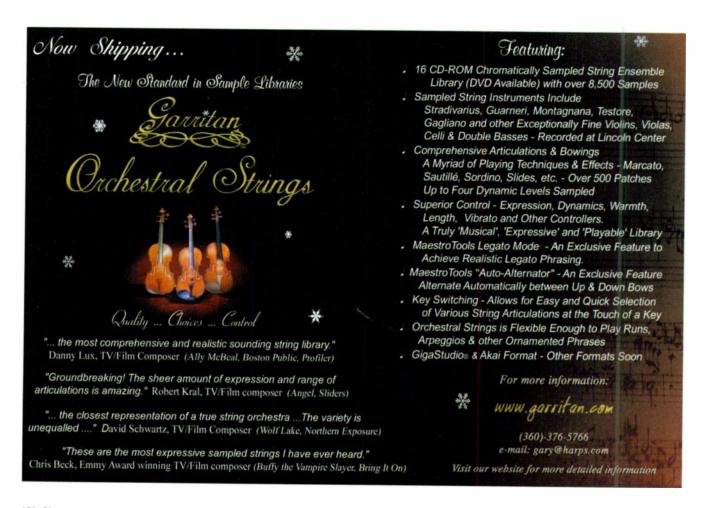
### LAY IT OUT

*Spektral Delay* is laid out in a fairly straightforward manner. All controls are presented in one large window, with the three main parameter controls (Filter, Frequency, and Feedback) occupying much of the space (see **Fig. 1**). The Modulation controls appear along the top of the window and the file-related ones along the bottom. Gain and Wet/Dry Mix controls are in the upper left corner, as is a preset-management section for storing and recalling user-created parameter presets.

The program's three main controls are presented as Matrix Editors, which are rectangular edit windows where horizontal *(x)* and vertical *(y)* mouse movements control two independent parameters. In the Filter editor, *x* controls the amplitude of the frequency selected by the *y* movement. In the Frequency editor, *x* is delay time and *y* is frequency, and in the Feedback editor, *x* is feedback amount and *y* is frequency.

Each of the three editors provides a left and a right pane for stereo signals, but they can be unlinked to allow independent processing of each channel for each matrix, which is crucial. Some of the most interesting results came from creating a stereo-like effect by using subtly or drastically different settings for the left and right channels in

any or all of the three editors. Each editor can also be bypassed, if desired.

The program's overall layout makes sense but can be a bit hard to get used to. That's because the horizontal *(x)* values increase as the cursor moves from left to right, which causes a solid graphic fill to be displayed in the cursor's wake. However, the background behind the fill is a striped pattern, which plays a trick on the eyes and can be confusing: it looks as though the background is actually the fill. I wish that the background and fill patterns were swapped for the sake of clarity.

To the left of the main controls is the Input Modulation section, which offers several preset modulation algorithms that further mangle the sonic transformations controlled by the rest of the program. The algorithms have colorful names such as Deterioration, Jello Mold, Phase Blaster, Pitch Roll, and Lime Twist, and make for good starting points (the manual describes the function of each algorithm). I particularly liked Phase Blaster for its watery, shifting quality and Deterioration for its mangling effect. (You can hear examples of the algorithms at the **EM** Web site.)

The Input Modulation section provides three knobs for independently controlling the parameters of the left and right channels. The knobs can be linked, and their functions change with the selected algorithm.
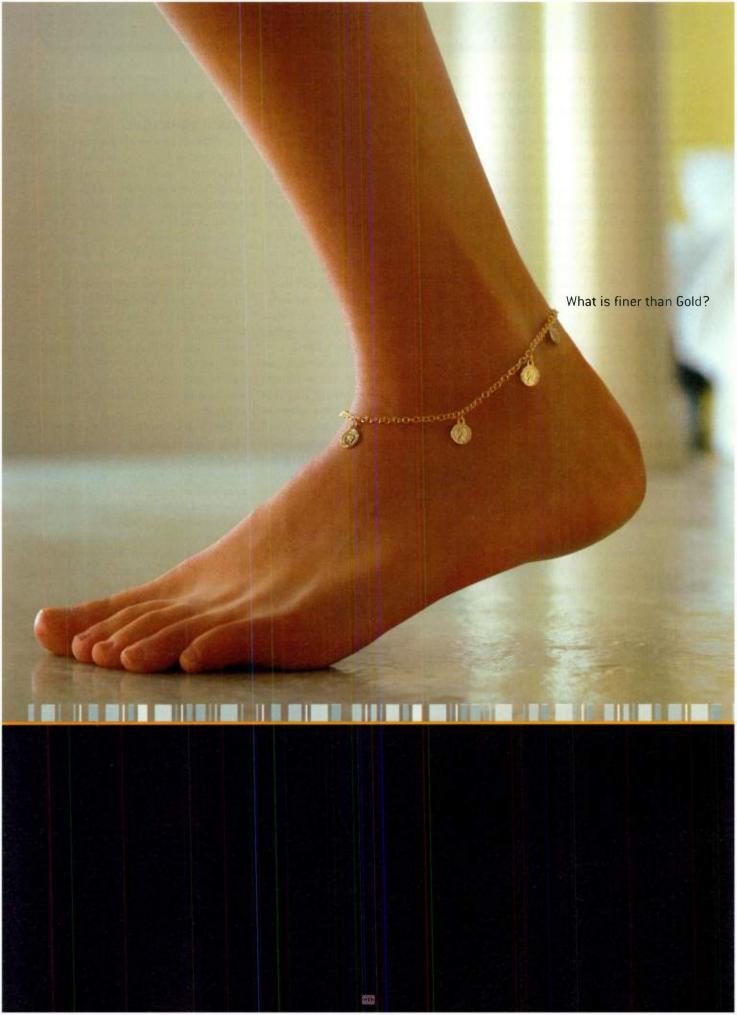
At the extreme left and right sides of the main window are the input and output Sonogram windows. Although not essential to the program's operation, those displays provide a continuously scrolling graphic representation of the spectral content of the sound going through the program. Native Instruments says the Sonogram windows don't impact the program's performance.

### SPEKTRAL COMPONENTS

You'll find other transformation possibilities in the Editor panel, which lets you apply mathematical functions to the various edit matrices (see **Fig. 2**). For example, the Quantize function, which applies only to the Delay matrix, allows edit gestures to be quantized to the current tempo grid (if one is active). Other functions include sine, triangle, sawtooth up, sawtooth down, and random wave shapes. Each shapes the selected area of the matrix being edited if a selection exists or the whole matrix if no area has been selected. Selections are made using the Marquee icon next to *Spektral Delay's* pencil tool.

*Spektral Delay* also has a single LFO that you can apply independently to

What is finer than Gold?

three destination parameters simultaneously. Each LFO instance can have a different modulation depth. The available destination choices cover all possibilities of x and y modulation of each of the three matrices, so many radical effects are possible. For instance, setting the LFO to a slow sine wave and modulating high-frequency attenuation and low-frequency feedback positively while modulating midband delay time negatively yields a complex shifting timbral result on virtually any sustained sound. The effect is rather like an instant sound sculpture. Available LFO wave shapes are the same as the wave functions.

The Delay Matrix offers a range of maximum delay-time settings, as long as 3, 6, or 12 seconds. Those require 5, 9, and 17 MB of RAM respectively. The number of analysis bands you use determines the maximum resolution for each setting. Five settings control the number of analysis bands: 64, 128, 256, 512, and 1,024. Higher settings require more CPU power. However, more analysis bands do not automatically produce the best sound; the results depend largely on the sound being processed.

*Spektral Delay*'s master tempo can be synced to the tempo of a VST host application when running under VST. In addition, the three matrices can be automated using MIDI continuous controllers and will generate controller data when you tweak their controls manually. The program's MIDI implementation is thorough and allows everything from Effect Bypass to Mod Source Select to remote control of the matrices' settings. The comprehensive set of MIDI options adds a whole new dimension to the program's overall usability. Simultaneously automating such varied parameters as the Filter Matrix, LFO Mod Depth, and Feedback levels independently of each other, particularly with a sequencer, produces some amazing results.

### INS AND OUTS

The File Player, found below the Input Modulation section, is handy. It allows you to load a preexisting sound file (WAV or AIFF on both platforms and Sound Designer II on the Mac) into RAM and play it through the program in One-Shot or Looped mode. That was the first thing I tried when I installed the program—I was curious to hear how some of the samples from my sound library would be transformed. The results were interesting, though largely dependent on the source sample and the program settings.

*Spektral Delay*'s File Panel is intended to demonstrate the program's features through the use of sound files installed automatically from the CD. Every time I tried to invoke the File Panel, however, the program appeared to hang. Native Instruments says that was a known bug in version 1.0 that has been fixed.

Running the standalone version, you can record *Spektral Delay*'s output to disk using the File Recorder, found at the lower-right edge of the window. (In the plug-in version, you can route the output to your VST mixer or effects.) The Recorder supports the same formats as the Player. Like all *Spektral Delay* functions, the Recorder's performance was sluggish on my Power Computing clone. According to the manufacturer, you will get much better performance with a native Power Mac G3 or G4, thanks to the newer computers' faster RAM and system bus.

### SONIC BEHAVIOR

It is difficult to sum up the sonic quality of *Spektral Delay*, but some adjectives that come to mind include watery, psychedelic, otherworldly, and complex. As with any delay system that includes feedback and long delay times, it's easy to obtain spaced-out results. However, the program's highly controllable frequency parameters set it apart from the others. You could easily configure a process in which the middle frequencies arrive later than the highs while the highs feed back endlessly, for example. The ability to rearrange various frequencies of a sound over time and to alter their behavior makes it easy to animate even fairly static source material.

The modulation capabilities of *Spektral Delay* make it possible for nearly all of the program's parameter values to be constantly changing—it doesn't take much of an imaginative leap to see where that might lead. Although I found the character of the program's sonic texture somewhat similar to that of some Steinberg GRM Tools plug-ins, particularly the GRM Tools equalizer and filter, *Spektral Delay* really does sound like itself more than anything else.

Overall, *Spektral Delay* is an amazing tool. I found it easy to create endlessly varying psychedelia and soundscapes using a wide range of input sound material. The possibilities afforded by the editors in conjunction with the modulation and MIDI control capabilities are staggering. The File Recorder makes the program viable as a standalone application and is incredibly useful for capturing magic moments while twiddling parameters. Be warned, however: you'll be happiest using a truly fast machine; like all Native Instruments' software, *Spektral Delay* is quite CPU-intensive. Nevertheless, this one is a must-hear.

*Peter Freeman is a bassist, composer, and producer in New York. He has worked with such artists as Seal, John Cale and Chris Spedding, Jon Hassell, Nile Rodgers, Sussan Deyhim, and Shawn Colvin.*

# YAMAHA

## DX200

### *Old-school digital synth meets old-school sequencer.*

**By Marty Cutler**

When Yamaha introduced the DX7 in 1983, its crisp, detailed, and sometimes clangorous timbres provided stark relief to the cloudy, warm tones of analog synths. The DX7 and its ensuing frequency-modulation (FM) synthesis counterparts had a few drawbacks, however; compared with the capabilities of analog synthesizers, real-time control was minimal. The DX7 couldn't even come close to delivering the timbral warmth and thickness typical of analog synthesizers. By the time I sold my DX7 in the late '80s, the demand for FM synthesis was largely overshadowed by the clean, detailed timbres of sample-playback synthesizers.

During the past decade, that appears to have come full circle. Real-time control is again part of the musical toolbox. Analog synths (authentic and modeled) are again in favor in the electronic-music marketplace. Still, Yamaha has continued to champion FM synthesizers with a new wrinkle here and a new feature there.

Enter Yamaha's DX200, a groove box that derives its sounds from an FM synthesizer and a sample-playback engine. Don't be fooled into associating the unit with your old FM sound cards or even your antiquated DX7; the DX200 is a synth of a different stripe. In the DX200, the once cold, brittle sounds of FM synthesis have undergone a truly impressive transformation.

## COOK'S TOUR

The DX200's work surface is divided into several functions, many of which overlap (see **Fig. 1**). The Control section in the lower left corner contains transport controls and buttons to enable the keyboard, shift octaves, tap in tempo, add swing, and perform additional chores. The Shift button accesses secondary functions, which are stenciled in raised letters below the buttons or indicated by a slash next to a knob's main function. Hold the Shift button and turn the filter's Cutoff knob to select a different filter type, for instance.

Directly above the Control section, nine backlit indicators that denote the mode in which you're working accompany a four-character LED display. The

Store button saves your edited Voices, Patterns, and Songs. The Show Value button interacts with the synth voicing section by "freezing" your most recently edited synth parameter values.

The DX200 offers many thoughtful programming conveniences. If tweaking has taken you far astray of where you started, hold the Shift and the Show Value buttons simultaneously to display a parameter's original value. The Exit button unfreezes the synth parameter controls, and the Data knob changes values in conjunction with other controls.

To the right of the DX200's Control section is a 16-button keypad that serves as a single-octave keyboard for entering notes. Keys 1, 4, 8, and 10 don't transmit notes, which can be somewhat confusing in the midst of pecking out a pattern, but fortunately, the active keys light up. The Shift key engages the bottom row of keys in various Pattern- and Song-editing functions.

The Voice section dedicates 16 knobs and 16 buttons to controlling various



**FIG. 1: The DX200's surface is divided into four overlapping work areas. Transport controls are in the lower-right corner, and knobs for altering synthesis parameters dominate the center.**

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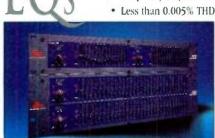
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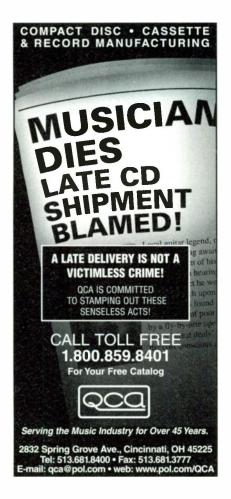
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sound parameters. Voice-editing capabilities make the DX200 really shine, so perhaps it's best to take a take a closer look at the synthesis functions before I describe the controls in greater detail.

### SMOOTH OPERATORS

Yamaha based the DX200's FM sound engine on the same 16-note polyphonic, 6-operator synthesis as the DX7. The DX200 arranges its FM Operators into the same 32 algorithms as its predecessor, and using System Exclusive (SysEx) messages, you can access the DX7's voicing parameters.

Overlaid on the FM synth is a host of subtractive-synthesis features that most significantly provide a powerful multimode filter. Its responses range from 24, 18, and 12 dB lowpass to bandpass, highpass, and band eliminate (see Fig. 2). All filter types are resonant, and the 24 dB model can be driven into whistling self-oscillation.

In addition to the six envelope generators (EGs) provided by the FM operators, the DX200 has ADSR envelopes for the filter and amplifier sections. An LFO (again, in addition to the one embedded in the FM engine) provides a choice of triangle, sawtooth (up and down), square, and sample-and-hold waveforms. You can assign the LFO to modulate amplitude, filter frequency, or pitch.

### BEYOND FM

The DX200's sample-playback engine offers 32 notes of polyphony. Instead of multiprogram Performances, the DX200 provides a single keymap that consists of three bass sounds occupying the greater part of the bottom three octaves and an assortment of drums and other sounds from MIDI Note Number 36 through 120. Unfortunately, each bass extends only 11 semitones before the next sound kicks in; that is definitely not the sound

set for Jaco Pastorius devotees. The drum sounds run the gamut from synthetic snares, kicks, and hats to processed- and natural-sounding acoustic drums and cymbals to udu and tabla variations. Higher on the keymap are thwacks, claps, smacks, and chordal instrument hits; some sound natural, and others, blatantly synthetic.

The drum and bass sounds are gated by the DX200's keypads or through MIDI. The inability to mute one sound with another (such as a closed hi-hat muting an open one) is doubly unfortunate, because you can't easily create realistic hi-hat parts. Cymbals cut short immediately after you release their triggers, making realistic cymbal parts difficult, if not impossible, to achieve.

In the Noise Oscillator section, a button lets you select from five noise types. You can sweep the noise source's filter frequency, but that's about it. Although the Noise Oscillator doesn't interact with the FM tone generators, FM synthesis can generate plenty of interesting noise on its own.

### IN REAL TIME

All of the DX200's 18 knobs (except the Main Volume knob) send MIDI Control Change (CC) messages. The Data knob sends messages based on its context. For example, when you hit the Pattern Select button, the Data knob sends Bank Select and Program Change commands. If you twirl the filter's Cutoff knob, the Data knob duplicates its function but with finer resolution. Both the Data knob and any parameter knob
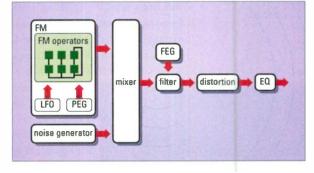


FIG. 2: The DX200 boasts 6-operator FM synthesis overlaid with subtractive-synthesis capabilities. Fully programmed FM sounds pass through the unit's multimode filter, which is controlled over time by the Filter Envelope Generator's (FEG) ADSR settings.

send a CC message beginning with the parameter's current value, preserving smooth timbral changes by preventing the unit from sending discontiguous values. Because the Data knob offers finer resolution, it sends data more fluidly than a parameter knob, which can achieve a stair-stepped effect if that's your goal.

The DX200's control surface provides a tremendous complement of real-time voicing capabilities. Four knobs, one each for Attack, Decay, Sustain, and Release, afford hands-on control of amplitude and timbre changes over time, with a button to switch between amplitude and filter envelopes. I would prefer four knobs for the filter envelope and four more for the amplifier envelope, but considering the small footprint of the box and the number of controls, a toggle button is a reasonable design compromise.

The filter section sports three knobs for cutoff, resonance, and envelope depth. By adjusting cutoff and resonance settings, I was able to make the filter oscillate. When I applied the EG to modulate the filter, I created a melody that followed the pitches in a Pattern I was playing.

With the Shift key, the Cutoff knob selects among filter types. You can invert the filter's envelope with the Filter Envelope Generator (FEG) Depth knob—a nice touch. Depending on cutoff and resonance values, settings at either extreme of the envelope depth can overdrive the filter, creating short bursts of howling feedback that sound great.

The DX200's effects section is rather spartan, but that's a minor quibble, considering the unit's price. Chorus and reverb supplement a modest assortment of delays, flangers, phasers, and amp simulators. Sparse though they may be, you can sync time-based effects to a song's tempo or MIDI Clock. For the delays, flangers, and phasers, the Parameter knob controls rate, and for overdrive, it controls distortion depth. The effects processor doesn't provide multi-effects, so you can choose only one effect at a time. The selected effect applies to FM and sample-playback tones alike.

## DX200 Specifications

| | |
|---|---|
| **Synthesis** | 6-operator digital FM; sample playback |
| **Maximum Polyphony** | 16-note (FM); 32-note (sample playback) |
| **Multitimbral Parts** | 1 (FM); 3 (sample playback) |
| **Effects** | delay, reverb, chorus, flange, phase, overdrive, 2-band EQ |
| **Sequencer** | 16-step, analog style with Free EG to capture CCs |
| **Analog Outputs** | (2) unbalanced ¼"; (1) ¼" stereo headphone |
| **MIDI Ports** | In; Out |
| **Display** | 4-character, 7-segment LED |
| **Dimensions** | 13.50" (L) × 2.20" (H) × 8.25" (D) |
| **Weight** | 3.5 lb. |

### ON THE FM DIAL

The transformations that occur when you can apply real-time controls to an FM synth are a revelation; the DX200's Harmonic and FM Depth knobs are especially effective. The Harmonic knob changes the waveform by continuously varying the frequency of the FM modulators. For example, I selected a rich sawtooth pad and adjusted the Harmonic knob to smoothly change the sound from a muted, brassy character to a bright and bell-like timbre rich with inharmonic overtones.

The FM Depth knob controls the amplitude of the modulators, which allowed me to change a bland sine wave to a harsh, metallic, distorted timbre with overtones that an analog synth could never produce. Using the second Decay knob, which controls only the FM modulators, you can vary the sound from sustained tones with overlapping tails to short, clicking tones. Like the filter envelope, you can invert the modulation values of the Harmonic, FM Depth, and Decay knobs.
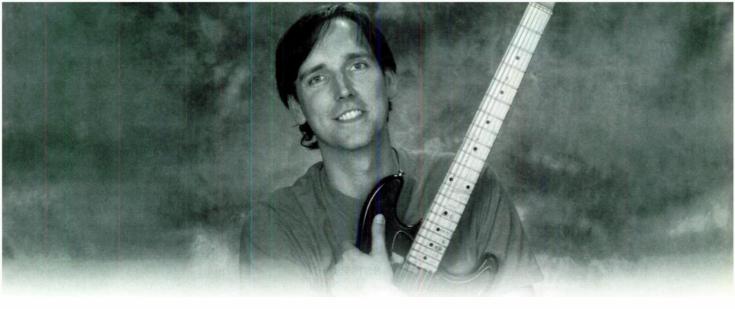
Judicious use of the Modulator button paints your sounds with a finer brush by controlling all the modulators or any one of three. By choosing a single modulator, you can make subtle changes to a single element using the Harmonic and FM Depth knobs. A 6-operator FM patch can contain more than three modulators, but given the wealth of timbral modifiers versus the available surface area, I can accept a few design compromises; I don't want to be greedy.

The Algorithm Type button toggles through all 32 FM algorithms on the fly, drastically altering timbre. Repeated button presses call up the next algorithm in line, from high to low, and then loop back to your initial selection. Pressing the button rhythmically gives the sound a Korg Wavestation–like motion. To scroll through a limited range of algorithms, you can use the Data knob, but for creating rhythmic effects, a button works better. I was disheartened to discover that I could not record algorithm changes into the DX200's sequencer, but I easily obtained the effect I wanted by recording to my computer's sequencer.

### ASSEMBLY LINE

Most of the DX200's sequencer controls are simple and mainstream. As with most pattern-based sequencers (and most drum machines), you create Patterns and link them together into a Song. Patterns are limited to single measures of 8, 12, or 16 steps.

The DX200's song-assembly style leaves much to be desired; constructing music one measure at a time is counterintuitive. Even modern hardware sequencers let you assemble parts of a composition from start to finish without the need to link patterns. Furthermore, the keyboard sends a fixed Velocity message, which you can adjust only by pressing and holding a combination of buttons. Nonetheless, the DX200's sequencer offers a bunch of cool tricks all its own, including a

# "Four Major Labels Came to See Me Because I Joined TAXI"

*Lizard McGee -- TAXI Member*

**M**ost musicians never get a chance to meet an A&R person in the flesh. I had A&R guys from Columbia, Dreamworks, Maverick and Hollywood all come to see my band, Earwig, play live.

I spent the next day hanging out with one of them at his house. I played more songs, and we talked one-on-one for hours.

All this happened as a direct result of becoming a member of TAXI.

Ironically, I almost didn't join. Like so many other people, I didn't know a lot about TAXI, and I wondered if it was really legitimate. It just sounded too good to be true.

But I spoke with a few friends who were already members, and they explained how TAXI worked. It made sense.

I began to think about not only getting my music to record labels and publishers, but also pitching my songs to TV shows and movies to make some extra money with my music.

So, I joined, and it's already paying off big-time. Earwig is building a huge buzz because of all the contacts we've made through TAXI.

We haven't signed a deal yet, but we've definitely penetrated the so-called "inner circle" of the music industry. And that's exactly where you need to be to get yourself signed.

Can TAXI get you into the inner circle? They'd be the first to tell you they can't promise anything. But four A&R people watching my show was all the proof I needed to know that TAXI can really deliver, if your music is right on target.

And if your music is a little bit off-the-mark, TAXI is probably the best thing you can do to whip it into shape. The written feedback you'll get from their A&R department is incredible.

You'll also get to meet top industry executives face-to-face at TAXI's annual convention, the Road Rally. As a member, you'll get FREE passes for you and a couple of guests.

This private convention is renowned for being the best in the business. Just one pass is worth far more than your TAXI membership fee, but you'll get three for FREE.

Whether you're pitching yourself as an artist, pitching your songs, or going for Film and TV placements, TAXI is definitely the place you need to call.

Just ask for their free information kit, and get yourself signed up in a hurry.

I did, and my only regret is that I didn't do it sooner. TAXI has turned out to be the best investment I've ever made in myself.

continuously variable Swing function and a button that instantly reverses the pattern.

The sequencer's star feature, however, is the Free EG section. Free EG captures and plays back knob and switch moves, thereby re-creating whatever motion you impart to the synth in real time. The DX200 has four Free EGs, and their lengths can be set to overlap Patterns. Free EGs can be triggered when you press the Play button, send a MIDI Start message, or send a Note On. Note On triggering animates the DX200's sound, even if you use it as a passive sound module.

Free EG can capture most knob and switch movements, but I was disappointed to find a few movements it doesn't record. I was delighted when I started playing rhythm patterns as I tuned the filter cutoff for a wah-wah effect, but when I tried to overdub my filter tweaks, I quickly discovered that the filter section is disabled when you record the sampled parts. Similarly, Algorithm Type selection is disabled when you record FM synth patterns. The Cutoff knob and the Algorithm Type button send MIDI CCs, so you can record them into an external sequencer, but I'm disappointed that the internal sequencer lacks that capability.

The DX200 also lacks a Pitch Bend control. You can certainly send Pitch Bend from an external source, but because the DX200 is so self-contained, I'm surprised that that basic feature was not assigned to a knob.

Perhaps the strangest limitation is that the sequencer track that records FM sounds is monophonic. Adding a multitimbral FM section would probably have increased the unit's cost, but I can't understand why a 16-note polyphonic synth should be hobbled by a monophonic sequencer part.

## OPERATOR ASSISTANCE

The manual is well written and includes 93 detailed tips for using the DX200. If you're new to FM synthesis, the appendix gives you a good, nontechnical introduction to the underlying concepts.

Yamaha bundles a *DX200 Editor* program for Windows and the Mac OS that allows you to edit and store sound parameters and sequences. You can construct an entire song on your computer and then dump the data to the DX200. The software provides a sequencer section that resembles an analog-style sequencer, with virtual knobs for setting the pitch of each step (see **Fig. 3**). You can even draw parameter changes into a window that displays the Free EG.

With *DX200 Editor*'s Voice-editing facilities, you can graphically get down to the FM nitty-gritty and then polish your FM sounds with subtractive synthesis parameters. The Voice-editing section features a few parameters that aren't accessible from the hardware, including a 2-band EQ.

A DX7 Simulator window displays a virtual DX7 keyboard for triggering sounds and editing Voices. The simulation is comprehensive, down to its virtual membrane switches and function buttons. Clicking on any button reveals a close-up of the Edit window. Although the simulator lets you edit FM Voices as though it were a DX7, why would anyone want to? Thanks, but I've done that already.

## IS IT GROOVY?

The DX200 is a blast to play. The synergy of FM synthesis and subtractive synthesis coupled with real-time control is fresh and powerful. Because the DX200 is compatible with DX7 SysEx, an enormous collection of ready-made sounds is available, and each Voice can seed tons of variations.

I'm not too enthusiastic about the sequencer section, which is a curious mixture of real-time flash and unwieldy button presses. I'd rather sequence a song from beginning to end than arrange single-measure patterns into song form. The sequencer's lack of polyphonic FM playback is a drag, and the keypads aren't Velocity-sensitive. I'm disappointed at the omission of certain real-time recording features, such as changing algorithms on the fly. Devotees of analog-style step sequencing might find a lot to like, but I don't.

On the other hand, strictly as a sound module, the DX200 works well. Clearly, the DX200's designers strove to present an analog, subtractive surface on the DX200, but that's only the tip of the tonal iceberg. The DX200 is quite capable of creating effective analog-type sounds. The fat, resonant filters impart a warm quality to the typically brittle sound of digital FM. The DX200's synthesis capabilities far outweigh its utility as a groove box. Give it a thorough listen and spin some knobs.



**FIG. 3: The *DX200 Editor* software lets you create Patterns and Songs on your computer. The user interface resembles an analog step sequencer, with virtual knobs for note, gate time, and Velocity.**

*Assistant editor **Marty Cutler** is known for his "outside" approach to bluegrass banjo: his parents always told him to play there.*

# TC ELECTRONIC

## TRIPLE-C

### A rackmount digital compressor with a few new twists.

**By Rick DiFonzo**

Digital compressors have never been highly regarded. Despite their obvious advantages over analog compressors—automation, more precise control, and program storage (thus repeatability)—many recordists feel that they just don't sound as good. Fortunately, as with all things digital, the technology has improved by leaps and bounds, and the bad rap for digital compressors might finally be on the wane.

TC Electronic's latest digital signal processor, the Triple-C multiband compressor and envelope (see **Fig. 1**), could well have the right stuff for winning over even the most die-hard analog curmudgeon. In addition to full-range dynamics processing, the voltage-controlled-amplifier-based Triple-C provides two additional compression modes, a sidechain function, and several new music-enhancing features not typically found on compressors. As for repeatability, the Triple-C holds 50 factory presets in ROM and an additional 100 user presets in RAM.

The Triple-C comes in two versions: stereo and mono. For this review, I tested the mono version in my personal studio.

## REAL HARDWARE

The Triple-C's front panel is logically laid out in six sections: Input, Multispectral LCD, Dynamic, Spectral Levels, Makeup Gain, and System. The Input section provides a Level knob with decibel markings at –6, 0, +3, +6, +9, +12, and +18.

The Dynamic section provides control of the usual compressor parameters through four dedicated, continuously variable knobs labeled Threshold, Ratio, Attack, and Release. The Threshold knob has decibel markings at –40, –30, –20, –15, –10, –5, and 0, and the Ratio knob has markings for 1:1, 2:1, 3:1, 4:1, 6:1, 8:1, and ∞:1 ratios. The Attack knob has markings at 0.2, 0.7, 2, 5, 10, 20, and 50 ms. The Release knob's millisecond markings are at 20, 100, 200, 500, 700, 1k, and 2k. Also in the Dynamic sections are four buttons labeled Multiband Off, Peak Sensitive, Softlim, and Look Ahead.

The Spectral Levels section contains two detented knobs labeled Lo-Band and Hi-Band, each with markings for Min (counterclockwise), Normal (straight up), and Max (clockwise) settings. The section also has a button labeled Envelope Mode. The Makeup Gain section provides a Bypass button and a Level knob with decibel markings at –18, –12, –6, 0, +6, +12, and +18.

The System section comprises two concentric knobs and a Menu key. Use the outer Parameter knob to scroll between parameters and the inner Value Set/Enter knob to change values. You can also turn the Value Set/Enter knob to scroll through menu selections (after pushing the Menu key) or push it to approve actions such as recall and store.

The multispectral LCD is large, well organized, and pleasing to the eye. Divided into four sections, the LCD clearly shows all the information you need. On the left is a pair of vertical meters that display input and output levels. A prominent meter in the display's top-center section shows total gain through the unit and, when the unit is in Multiband mode, gain reduction for each band. That section also shows the input source (analog or digital), sync status, and sampling rate (44.1 or 48 kHz), and provides indicators for Link mode and MIDI receive. The Envelope window, located at the top right, uses diagrams to show the shape of the attack and release. A fourth section along the bottom of the display shows the patch name and corresponding parameter value in large type, which is nice considering how difficult it is to read the front panel's silk-screened labels.

The stereo Triple-C's rear panel provides two balanced ¼-inch TRS inputs; two balanced ¼-inch TRS outputs (the ins and outs also accept unbalanced connections); S/PDIF I/O on RCA jacks; MIDI In, Out, and Thru ports; and a ¼-inch external control jack for connecting a bypass pedal (see **Fig. 2**). The rear panel of the mono Triple-C differs slightly: in place of the stereo version's second analog I/O is a sidechain input and direct output (for linking two mono units). Both versions (stereo and mono) have an internal power supply and provide a standard IEC connector on the back panel for attaching the power cord.

## WE THREE MODES

The Triple-C's three modes of operation are Full-range, Multiband, and Envelope. In Full-range mode, the Triple-C functions as a conventional compressor. You can adjust attack times from 0.2 to 70 ms (not 50 ms, as the front panel indicates) and release times from 20 ms to 2 seconds.

The Triple-C provides a greater range of compression ratios than most compressors, ranging from 1.12:1 (not 1:1, as labeled on the front panel) to 64:1 and on to ∞:1. As you change the ratio, the values show in the display.



**FIG. 1: In addition to traditional compression, the versatile Triple-C offers Multiband and Envelope modes, tube simulation, 50 factory presets, 100 user programs, and extensive parameter control through MIDI.**

Multiband mode splits the input into three user-defined bands—designated Lo, Mid, and Hi—and compresses each band independently. In that mode, the threshold, ratio, attack, and release controls affect the Mid band. The other bands are compressed with the same ratio, attack, and release, but only if they exceed the threshold.

Spectral Levels boost or cut the Lo and Hi bands; think of Spectral Levels as makeup-gain controls for the Lo and Hi bands. If the one band is being squashed, you can use Spectral Levels' controls to bring its level back up a bit; they allow very precise tweaking and subtle shaping of harmonically complex signals.

Multiband mode also provides the Look Ahead feature, which delays the output by 3 ms. With Look Ahead engaged, the Triple-C analyzes the incoming signal, and if it sees a peak that exceeds the threshold, it squashes it before it appears at the output. Because most people don't perceive such a short delay, the attack time in Look Ahead is virtually 0 ms. (I wish this feature were available in Full-range mode.) One minor annoyance was that my test unit exhibited an audible pop every time I pressed the Look Ahead button.

Peak operation is another feature that's available only in Multiband mode. The Triple-C is basically an RMS-based compressor, meaning that it responds to the average level of the incoming source material. The Peak Sensitive switch lets you change from RMS to Peak operation, letting you use the unit as a peak limiter.

Envelope mode is unusual, if not unique, in that it lets you independently adjust the signal's attack and release times. Envelope mode makes it possible to shape a signal's dynamic content throughout its duration, which is pretty cool. I used it successfully to soften the pick attack on a bass-guitar track, accentuate the pick attack on a biting electric-guitar track, and dry up a wet signal somewhat by dipping the reverb tail.

The Envelope mode's effect was most apparent when I used it to add attack to a snare drum played with brushes. The range of settings let me soften the brush hits to mush, make them crack like a stick hit, and dial in any sound between those two extremes. Be careful when monitoring at high levels, though; with a fast attack and the threshold set near 0 (thus affecting all signals), attack transients can be sharp enough to damage your speakers.

## MAY I SEE A MENU?

The Triple-C's concentric knobs let you recall, edit, and store factory and user presets easily. To recall a preset, scroll with either knob until you find the desired patch and then press the inner Value Set/Enter knob. When you've finished editing the preset's parameters, select Store from the Edit menu. You then have the option of storing that patch in the first available User patch location. If you wish to overwrite one of the 100 User locations, just scroll to it and save. Before saving, the Triple-C prompts you to name the patch. You can archive User patches through a MIDI Bulk Dump.

Pressing the Triple-C's Menu button provides access to all menus. The Edit menu lets you change the crossover frequencies of the Lo and Hi bands, select patch categories (vocal, percussion, bass, horns, and so on), and enable or disable the unit's sidechain.

The I/O menu provides access to a slew of options. You can choose analog or digital input source, clock settings (44.1 kHz, 48 kHz, or external), AES/EBU or S/PDIF operation, digital input levels, analog output levels, and dithering alternatives. The I/O menu is also where you find the Triple-C's MIDI functions—MIDI channel, Control Change (CC) enable, Program Bank (where you determine whether CC messages call up factory or user presets), Bulk Dump, and SysEx ID—as well as the means to enable the Link function when chaining together two units.

The Triple-C's MIDI implementation affords access to parameters through Control Change messages. That useful feature helps when a single compression setting isn't sufficient for a whole track. For example, you could automate the unit with MIDI commands to provide really hard compression on the lead vocal during the chorus and gentle compression during the verses. Furthermore, you could automate changes in tonal characteristics by changing the Hi-Band and Lo-Band settings in Multiband mode.

The manual also lists Bypass and View Angle in the I/O menu, but those options were missing from early versions of the Triple-C, including the one I

## Triple-C Specifications

| | |
|---|---|
| Digital I/O | (2) S/PDIF |
| Analog Inputs | (2) balanced ¼" TRS |
| Analog Outputs | (2) balanced ¼" TRS |
| Additional Ports | MIDI In, Out, Thru; (1) ¼" TS footpedal |
| Sampling Rates | 44.1, 48 kHz |
| Output Dither | 24-, 20-, 16-, 8-bit |
| D/A Conversion | 24-bit, 128× oversampling |
| Display | 23-character, 280-icon Super Twisted Nematic (STN) LCD |
| Operating Level | +4 dBu |
| Dynamic Range | 100 dB (20 Hz–20 kHz; analog input) |
| Total Harmonic Distortion | 0.002% (@1 kHz, +20 dBu; analog output) |
| Frequency Response | 20 Hz–20 kHz (+0/–0.5 dB @ 48 kHz; analog output) |
| RAM/ROM Programs | 100/50 |
| Dimensions | 1U × 8.2" (D) |
| Weight | 4.1 lb. |

tested. If you happen to have one of those early units, TC will update it to the current software revision for free.

## CHAIN, CHAIN, CHAIN

The Triple-C's Sidechain function works like any compressor sidechain, letting another device—an equalizer, for example—control how the Triple-C behaves by triggering the compression.

When you select Add from the menu, the internal settings and the external device share control of the compression—equally, using a summing amplifier—which kicks in when the combined signal of the input and the sidechain exceeds the threshold. (I was disappointed that the Add function wasn't explained more adequately in the manual.)

You can also use the sidechain input to link two mono Triple-Cs for stereo operation. Connect the direct out of the master Triple-C to the sidechain in of the slave Triple-C and vice versa, connect MIDI Out from the master to MIDI In on the slave, and then select Link On in the I/O menu on both units. In stereo mode, both units respond to the signal present in the master unit. The master's threshold determines when compression kicks in on both units, providing operation identical to that of a stereo compressor.

You can also set up a dual-mono mode by making only the MIDI connection and again selecting Link On in the I/O menu. In dual-mono mode, you set the parameters for both units from the master, but each unit responds to its own inputs. Dual-mono and stereo mode are both useful for compressing two tracks identically. Unfortunately, because I had only one mono Triple-C, I was unable to test those functions.

## THE REAL WORLD

Like most studio owners, I have my favorite compressors for specific applications. Some have tubes, others have solid-state circuitry, and a few are computer plug-ins. For example, I typically use Daking 91579s on drums, bass, electric guitars, and vocals. The Bomb Factory Classic Compressors plug-ins—the LA-2A and 1176—also get a lot of use on drums and guitars. For recording acoustic guitars, I usually prefer the dbx 166. If I want to add just a little compression to a mix, I patch my Audio Design Compex Limiter across the stereo bus. (For a comprehensive tutorial about compression and characteristics of different types of compressors, see "The Big Squeeze" in the February 2001 issue.)

The Triple-C stacked up well against the other compressors in my studio, and it proved extremely flexible, working well on everything I tried. It has become my first choice—or it at least receives equal consideration with my usual standbys—for several applications, including bass, acoustic guitars, vocals, and percussion.

To test the Triple-C, I first recorded a variety of instruments and vocals with no EQ and compression. I then patched the Triple-C into a channel insert and played back my tracks to see how the Triple-C performed.

I started by dialing in factory presets and tweaked from there. With Fast Bass (a Multiband preset), fiddling with only the threshold and makeup-gain controls produced desirable results for a pounding rock piece. Tube Bass Comp worked well right out of the box. For electric guitar, I chose the Natural GTR preset with no alterations other than slight adjustments to the Spectral Levels, which helped place the guitar track in the mix.

Heavy guitars benefited from a generous dose of Multiband compression with a slow attack time, especially when I boosted both Lo and Hi bands a bit, leaving a small notch in the middle. I

also achieved a very nice U2-like guitar sound using the Light GTR Comp patch with a fast attack time and a high ratio. Background Vocs added silkiness to layered vocals and made me wish I had two units to process the voices in stereo.

I tried cutting some tracks with the Triple-C in the signal chain. I plugged in my Bass Pod's balanced output and turned off the Pod's compression. The bass guitar growled nicely, and the Triple-C smoothed out its sound. Loud passages were held in check, and quiet passages exhibited both pick attack and finger noise—just what I wanted. Also, thanks to the Multiband compression containing the boomy, low-end stuff, I was able to print the signal hotter to disk.

**FIG. 2: The Triple-C's rear panel offers S/PDIF I/O, balanced analog I/O, a footswitch jack, and a trio of MIDI ports.**

All of my instrument tracks were improved by various amounts of compression ranging from subtle to over the top, especially in Multiband and Envelope modes. Generally, the factory settings were fine starting points for further tweaking, but because every compression task calls for its own settings, the ability to store 100 user-programmable patches is extremely useful.

## RADIANT HEALTH

One of the Triple-C's most outstanding features is the Digital Radiance Generator (DRG; accessible through the Edit menu), which adds second-harmonic distortion to the signal to simulate tube warmth. On acoustic guitar, the effect was subtle but nice; it added a richness to the sound that I really liked, and afterward the track blended better with the other tracks. The DRG is also a compelling effect for background vocals, adding a nice airy quality.

Because the DRG is under MIDI control, you can process different sections with varying amounts of "radiance" during mixdown. Although I didn't have access to a tube compressor for comparison purposes, I really liked the DRG's sound.

## MANUAL LABOR

I have a few minor complaints about the Triple-C's user manual. In some places, the writing is difficult to understand because of poorly constructed sentences and translation errors. For example, when describing the Multi-band mode's parameters, the manual reads, "By dividing the source material into three frequency areas you can avoid that peaks at certain frequencies controls the compression of the entire signal." Huh?

Another problem is that the manual's front-panel diagrams are difficult to read. Although the manual includes a brief, helpful introduction to compression and basic guidelines for novices, it inadequately explains or arbitrarily omits some features.

## CHRONICALLY COMPRESSED

The Triple-C is a reasonably priced, flexible, great-sounding digital compressor that is at home in pro and personal studios. Its three modes offer functionality I haven't seen in one box before. The Triple-C performs well in Full-range mode and merits consideration as the primary compressor for any decently equipped studio for that mode alone. In Multiband mode, however, the Triple-C really distinguishes itself. On top of that, you get Envelope mode and the DRG— quite a lot of truly useful features.

In the end, the ultimate test of a piece of gear is, will I buy one? I already own six compressors; do I really want this one too? Yes, without a doubt.

---

**Rick DiFonzo** *is a guitarist, composer, producer, engineer, programmer, and father. He has played with Bob Dylan, Mick Jagger, and Roger Waters, among others. Like other guitarists, he always needs something to whine about.*

# NEUMANN

## KM 183

*An omnidirectional mic with diffuse-field equalization.*

**By Myles Boisen**

I n the beginning, there was omni-directional: all microphones start out as omnis and are tweaked to acquire various directional characteristics. As the science of making microphones progressed (aided in no small part by Georg Neumann), unidirectional designs found favor for P.A. systems, broadcast, and music recording.

Thankfully, Neumann has continued to design and offer omnidirectional mics, including benchmarks such as the famed M 50. Now the venerable mic manufacturer has updated the Neumann KM 83 (the omni sibling to the well-known cardioid KM 84), logically christening the latest family member the KM 183 (see **Fig. 1**).

## WINDOW RATTLER

All KM 180–series Neumann mics are small-diaphragm, phantom-powered condensers employing transformerless FET electronics. At just over four inches in length, the transducers are extremely light and easy to position. All 180-series mics are manufactured in nickel and matte-black finishes, and accessories are available from the manufacturer. Although it appears that the threaded microphone capsules, which screw on to the preamp bodies, can be swapped (as is the case with the modular 100-series Neumanns), the 180-series capsules are not interchangeable.

Generally, small-diaphragm omni-directional microphones are favored for critical recording applications in which flat frequency response, accurate transients, and neutral, uncolored sound reproduction are prime concerns. For classical recording, film

sound, and sampling, that transducer class is highly regarded. In contemporary pop recording, however, omni mics are more of a specialty item. That is partly because of the dominance of close-miking techniques and perhaps also because of a prevailing notion that omni mics don't offer adequate low-end response.

Regarding the latter view, it's true that small-capsule omnis provide neither the enhanced midbass warmth of a large-diaphragm mic nor the proximity-dependent bass boosting inherent in all focused-pattern microphones. However, most quality omnis, including the KM 183, offer low-end response that is flat to 20 Hz and below. That translates to window-rattling lows when recording orchestral bass drums, organ pedals, gongs, acoustic and electric basses, and thundering rock drums (with the omnis used as room microphones).



**FIG. 1: The Neumann KM 183 has a presence boost that helps it sound crisp on distant sources, making it distinctive among small-diaphragm omnidirectional microphones.**

## HELPFUL BOOST

Another aspect of the KM 183's frequency response deserves a note of explanation. It is technically possible to design an omnidirectional microphone with ruler-flat frequency response beyond the audible frequency range: the Neumann KM 131 is such an item, as are models by DPA (formerly Brüel and Kjær), Earthworks, and others. But the neutrality of a flat microphone is not always desirable and can result in washed-out sound and lack of high-end detail, especially in concert-hall and studio recordings.

To compensate for the high-end loss that occurs when miking with an omni at a distance of three or more feet, Neumann designed the KM 183 to have a rising presence boost similar to that of its ubiquitous vocal mics. *Diffuse-field equalization*—a smooth increase in high-end response above 4 kHz, peaking with a 7 dB boost at 10 kHz—allows the KM 183 to sound crisp on distant sources.

That characteristic also lets the KM 183 compete in the studio as a close-miking tool, because it can enhance detail yet remain free of the muddiness that often attends bass boosting caused by the proximity effect. Because omni mics exhibit no proximity effect and off-axis coloration on high frequencies, they are surprisingly easy to position on complex instruments. Omnis are championed by a growing number of engineers for recording acoustic guitars, for example. Like other omnis, the KM 183 can be placed directly in front of a guitar's sound hole to capture rich resonance and plenty of high-end sparkle with little of the boominess and unfocused lows that afflict cardioid mics in that position.

## NO MATCH MADE IN HEAVEN

For this review, I received a pair of matte black KM 183s packaged as a set complete with swivelmounts and foam-rubber windscreens in a single wooden jeweler's box. Although the KM 183s are sold both individually and in stereo pairs (with sequential serial numbers), the stereo pairs are neither provided with frequency-response graphs nor

sonically matched. According to Sennheiser and Neumann product manager Karl Winkler, the caliber of construction and the quality-control process for Neumann mics ensure that any two mics of the same model will sound enough alike that, in general, they should work fine for stereo-recording applications. But again, no extra testing is done to precisely match the stereo pairs.

In loudspeaker tests with a two-way speaker system playing full-range rock mixes, I was surprised to hear distinct differences in frequency response between the two KM 183s. For the signal path, I used Baltic Latvian Universal Electronics (BLUE) Kiwi mic cables and a Langevin Dual Vocal Combo or Focusrite Green preamp, with gain matched by meters and by ear. All signals were recorded to DAT.

I initially positioned the KM 183 pair about two feet from the speaker, with the capsules approximately one inch apart. But I ended up moving the mics back to a distance of six feet from the source to minimize discrepancies caused by the relative proximity of the tweeter or woofer. At that distance, listening sessions in the studio (and later

in my mastering suite) revealed marked dissimilarities in presence and in high-end reproduction between the two mics and subtle differences in bass pickup. I tried switching both mic preamps and mixer channels but received the same results.

Specifically, one KM 183 had a noticeably bigger sound; it enhanced the mix's high-frequency components such as hi-hat, snare, and vocal sibilance, and it added a bit more punch to the kick drum. I also heard minor variations in the lower midrange, which affected the timbre of electric bass and distortion guitar in the mix.

I also recorded an acoustic guitar with both mics and compared the results. One mic was obviously brighter, and the other had a thicker, more woody-sounding midrange quality. However, both microphones sounded good, and the differences were not as pronounced as in the loudspeaker tests. The mics' output levels were closely matched too. But still, the KM 183s were not sufficiently matched for critical stereo recording. Were I to audition a pair for the purpose of buying just one KM 183, the obvious sonic differences would prompt me to do a lot of testing before deciding which flavor I preferred.

## DISTANT DAZZLER
To test the suitability of the KM 183s for classical-style recording, I set them up as a spaced pair to record composer Dan Plonsey's Portcullis Ensemble. Incidentally, during that session, I also tested an MBHO MBNM-622, a unique stereo mic that employs boundary-layer mics on either side of a modified Jecklin disk (see the review in the October 2001 issue).

In general, the KM 183 pair conveyed a realistic, unexaggerated left-to-right balance with a blended center image and an authentic-sounding ambience. Although each recording system had its strengths, the Neumann pair timbrally was the more natural, displaying open, airy detail that was particularly appealing when monitored on a pair of Grado SR 125 headphones.

Compared with the sharp separation

and drier quality of the MBNM-622 recording, the imaging of the KM 183 tracks was sometimes blurred with slight shifting in the perceived position of high-end instruments such as cymbals and piccolo. However, part of that could be attributed to room reflections (which are minimized by the MBNM-622) and to the aforementioned differences in high-end response between the KM 183 transducers.

Positioned about four feet in front of a Taylor acoustic guitar, a single KM 183 added a world of subtleties when mixed in with an XY pair of close mics (Oktava MK012s). Low-end fundamentals, pick sound, sparkling harmonics, and early reflections from the room were accurately represented by the KM 183, contributing an exciting, three-dimensional realism to what would otherwise have been a fairly standard foreground guitar track.

Throughout the tests and especially for ambient miking on quiet sources, the KM 183's low self-noise and relatively high output were a boon. The only hitch in that session came when the KM 183 faithfully conveyed some machine noise from another part of the building. A low-cut filter on the mic preamp effectively dealt with the unwanted rumble, and I made a mental note that industrial buildings and omni mics don't always get along.

I discovered another noteworthy use of the KM 183 as a distant microphone while recording the improvisational trio Maybe Monday. When cellist Joan Jeanrenaud (formerly of the Kronos Quartet) entered as a guest artist, I decided to isolate her in the drum room, away from the trio's electric guitarist, Fred Frith, and saxophonist, Larry Ochs. I left the door to the wood-floored drum room open and supplemented Jeanrenaud's close mic (a Royer SF-1 ribbon) with a KM 183 taped to the floor a few feet away.

Employing the KM 183 as a boundary-layer mic added some interesting and rather dark live reverb to the mix. The mic also picked up amazingly clear artifacts from Frith's amp, which was 30 feet away, around a corner. Mixing that room track in with the close-miked

# firewire
## storage is here!

cello track contributed a pleasant high end to arco passages. Used in moderation, that track filled in the instrument's sound transparently without adding boominess or boxy coloration.

## CLOSE ENCOUNTERS

In addition to being a reliable, smooth-sounding performer in conventional omni applications (distant miking of instruments and ensembles), the diminutive KM 183 also has some big surprises to offer as a stand-in for standard cardioid-pattern close-miking duties. For a session engineered by my studio partner, Bart Thurber, I proposed using the KM 183 as a single mic on a Hammond M-3 organ and Leslie Model 51 combination. At a distance of four feet and routed to a vintage Telefunken V72 tube preamp, the KM 183 delivered an even, airy sound with just enough ambience to grant the organ a spacious character. The Leslie cabinet had no problems with harshness or mechanical noise, and positioning the mic between the high and low rotors yielded plenty of low end. The only drawback was that the Leslie's speed control was installed on a homemade metal switching box, and even from ten feet away, the KM 183's big ears picked up the click of the switch.

I used the two KM 183s as a spaced pair on a set of three conga drums while recording rhythm tracks for a CD by singer and songwriter Jen Faith. Although I never tried that with omnidirectional mics before, after a couple of quick positioning adjustments, the Neumanns allayed any reservations I had. The low end of the conga set was rich in tone but never boomy, there was plenty of slap and skin sound, and the overall tone was tightly focused with a perfect amount of organic room sound.

Encouraged by those results, I threw caution to the wind and used one of the KM 183s during a session with jazz drummer Milo Francis—not as an overhead microphone, but in front of Francis's double-headed bass drum.

▼

### The KM 183's low self-noise and relatively high output were a boon.

When positioned as close as I usually like—one to two inches away, aimed at the outer rim of the head—the +25 dB gain setting on a Neve 1272 mic preamp provided way too much gain for my analog recorder. The mic has no attenuation pad, so I simply moved it back about four inches from the head and slightly lowered the preamp trim setting. The KM 183 never complained, and after a few EQ cuts in the low end and a small boost at 4 kHz, I had a solid, present bass-drum sound that provoked enthusiastic comments throughout the mixing session. I was knocked out by the KM 183's ability to capture that elusive jazz boom and appreciative that the leakage from the kit, though a bit more pronounced than with the usual cardioid mic, was even and uncolored.

## PLAIN SOUNDS GREAT

As an engineer who savors the rich qualities of room ambience in studio recordings, I have developed a great appreciation for omnidirectional microphones over the years. Some in my collection will do something to enhance room sound or contribute a unique equalization curve to close-miked sources. I have also tried ultra-flat instrumentation omnis in the studio but have generally found that they lack character for my preferred recording style.

Excluding the top-dollar mics, nothing that I've heard in the omnidirectional class compares to the pleasant neutrality and sweetened high end of the Neumann KM 183. Purists may debate the merits of diffuse-field equalization, but my ears reveal that this little mic has an amazing reach, and its high-end compensation delivered astounding nuances from near or far without ever sounding overly bright or fizzy.

As with all Neumann KM 100–series mics, low noise, lifelike transient response, and clear resolution of delicate acoustical details are hallmarks of the KM 183. The pair I tested was not sufficiently matched for use as a stereo set, which was a disappointment given Neumann's stature and reputation. Nonetheless, the mic just plain sounds great, and it should provide years of sonic revelations in the studio and for live recording. I highly recommend the KM 183 to those looking to branch out into ambient miking, expand their miking techniques, or simply upgrade from budget omnis.

**Myles Boisen** *is a guitarist, a producer, a composer, and the head engineer and instructor at Guerrilla Recording and the Headless Buddha Mastering Lab in Oakland, California. E-mail him at mylesaudio@aol.com.*

## KM 183 Specifications

| Element | externally polarized (DC bias) capacitor ("true" condenser) |
|---|---|
| Diaphragm | ½", 3-micron, gold-vapor-deposited Mylar |
| Polar Pattern | omnidirectional |
| Frequency Response | 20 Hz–20 kHz |
| Dynamic Range | 124 dB |
| Sensitivity | 12 mV/Pa (±1 dB; @ 1 kHz into 1 kΩ) |
| Signal-to-Noise Ratio | 78 dBA |
| Self-Noise | 16 dBA |
| Maximum SPL | 140 dB (for 0.5% THD) |
| Power | 48V phantom (±4V) |
| Dimensions | 4.21" (L) × 0.87" (D) |
| Weight | 0.18 lb. |

# ALGORITHMIC ARTS

## SOFTSTEP PRO 2.06A (WIN)

### *A mind-bending software music machine.*

### By Peter Hamlin

A fascinating program from Algorithmic Arts, *SoftStep* responds to your playing in creative and spontaneous new ways. Play a major scale on your MIDI controller, and the software responds with an exotic microtonal cluster. Play a few more notes, and it responds with a soft, delayed version of the melody at a tempo determined by your playing Velocity. When you pause for more than a specified period, a long arpeggiated chord sounds. The harmony unfolds in response to biological information from genetic code, and a background rhythm responds to the patterns of a fractal image on the screen. Those are only some of the musical scenarios you can produce with *SoftStep*.

Algorithmic Arts describes *SoftStep* as "a Windows-based, modular MIDI step sequencer and algorithmic-composing program patterned somewhat after the modular analog step sequencers: the big ones, with lots of knobs and blinky lights." On the Algorithmic Arts Web site, you can download a free introductory version called *SoftStep Basic*. There's also a fully loaded *Pro* version and *SoftStep LE* ($79), which is positioned about midway between the other two. The difference between them is the number of functions available. Although serious users will soon want to upgrade from the free version, *SoftStep Basic* provides more than enough for you to learn how to use the program and decide which level is best for your needs.

## FIRST STEPS

When you launch *SoftStep Pro*, you're greeted with an empty Workspace window. A menu bar at the top offers basic functions such as opening and saving files, creating modules, setting configurations, creating your own user functions, navigating the Workspace, and getting help (see **Fig. 1**). Below the menu bar are tools for creating a snapshot of a configuration and for storing and retrieving those settings. You can also launch the Composer utility, a playlist editor that lets you create a sequence of configuration snapshots. The Composer is useful for building large and varied compositions.

A metronome sets the master tempo, and the Run button acts as the Play control on a sequencer transport does. Monitor lights indicate MIDI input and output, quarter-note tempo ticks, and timing errors if they occur. You can show or hide a console of 16 faders. An indicator reminds you if the current configuration has changed since the last time you saved it.

## A REALLY BIG WORKSPACE

*SoftStep Pro*'s opening screen is just 1 of 16 work areas. The entire program is organized into four Banks of four Pages; the Banks are numbered 1 through 4, and the Pages are identified as North, East, South, and West.

The opening screen (Bank 1, North) is large enough for many typical configurations, but when you create a large, complex configuration that doesn't fit comfortably on one screen, you can span different Pages and Banks and switch easily between them. You select which Page is visible by clicking on buttons marked N, E, S, or W and toggling between the four Banks with another button. The program's modules can be interconnected across Bank and Page boundaries, so there are no limitations other than whether a Page is visible.

Pressing the Map button brings up an overview of all 16 Workspace areas, allowing you to keep track of the whole structure. You can click on any part of the Map overview to move to that Bank and Page. Such a system of organization works well: it avoids screen clutter and lets you organize your project any way you want.

## ALL THE INS AND OUTS

*SoftStep Pro* has 175 modules with various functions that you can freely interconnect. To create a module on the screen, left-click where you want the module to show. A pop-up list of categories arranged by function appears, and you can select the type of module you want from the list. Modules can also be selected from the Modules menu, or you can right-click in the Workspace to duplicate the last module you created.

Each module has one or more inputs, which appear as green rectangles for data and yellow rectangles for clock inputs. Clicking on an input produces a pop-up screen with a choice of numerical values, control flags, or the output



**FIG. 1: The main work areas in *SoftStep Pro* provide a series of menu options at the top and a large Workspace for building configurations in the remainder of the screen. This simple example shows a Clock module triggering a Counter module, which in turn cycles a sequencer through its eight steps. Values are set by the sequencer's knobs and are sent to the Note input of a MIDI-Voice module. A slider bar sets the volume.**

of any other module (*SoftStep* follows the MIDI standard of using numerical values 0 through 127). Rather than using visible "patch cords" to show connections, each input clearly displays the name of what has been connected to it. That keeps the screen clean and uncluttered.

Control flags indicate various system values and track information such as the Run button's status, which computer keys you press, or the status of Composer playlists as they run. For example, if you press a key on the computer keyboard, the KeyChar control flag will equal the ASCII code for that character, and that value will be available to any module that has KeyChar selected as its input. If you push the Run button on *SoftStep Pro*'s transport, the Run control flag will equal 127 (logical "true"). Pushing the Run button again will toggle it off, and the Run control flag will equal 0 (logical "false").

Having access to those system states

through the control flags provides many options for interactivity while the program plays. For instance, you could push the computer key that determines which chord the program will play, or you could simply use the Run button on the transport to start and stop Clock modules in a configuration.

When they're created, modules are assigned numbers to make them easily identifiable. For example, the first Clock module you create will be called Clock-1, followed by Clock-2, Clock-3, and so on. That makes the modules easy to differentiate when you interconnect them.

In my example of a simple *SoftStep* configuration in **Fig. 1**, a Clock module drives a Counter module, which causes an eight-knob Step-Sequencer module to



**FIG. 2: The Fill feature gives you powerful editing control over Step-Sequencer modules. Manipulating its graphic interface is easy; the data it creates can be used for many purposes.**

cycle through its eight positions. A MIDI-Voice module (labeled Voice-1) plays the notes specified by each knob setting on the sequencer. In other words, the Knob08-1 output controls the Note input of the MIDI-Voice module. The horizontal bar, marked 01 and set to 85, is used as a volume control and is connected to the Velocity input (Velo) of the MIDI-Voice module. Moving the

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right edge of the bar's blue rectangle changes the setting within the range from 0 (silence) to 127 (loudest). The Loudness Control information box identifies the horizontal bar's function.

### WATCH YOUR STEP

Many functions provided by the modules are straightforward, such as those that let you set values with a slider or knob. Other modules are more complex; the Kicker module, for example, monitors a stream of numbers and produces a clock pulse if less than a user-determined number of different notes appears within a user-determined period of time. You can use the Kicker to make sure there's enough variety in a sequence of numbers being generated in a configuration.

*SoftStep* provides Clock modules for timing; they include an innovative Ball module in which a ball creates interesting and varied rhythms as it bounces within four walls. You can adjust the box's size and shape and the ball's speed to get many variations in the rhythm. The Ball module creates a natural and changing rhythmic pattern without the rigid regularity of a precise MIDI Clock, offering one useful way to create more fluid and less mechanical timings in your configurations.

You can create rhythmic patterns using both the Arpeggio module and a series of Pattern modules. You can have as many as 128 separate patterns, each containing 32 steps, in a single module.

Central to the program are the many step sequencers, which offer different numbers of steps and distinct kinds of controls. The controls include knobs, sliders, and numerical inputs. Step sequencers with many steps work the same way the smaller ones do, but they take up more space on the screen; you are free to select the size you need. The individual controls allow you to enter data in the manner you find most convenient and most appropriate for the task.

Two other options are a choice of several types of Matrix modules, which are two-dimensional sequencers organized in rows and columns; and a choice of Page modules, which can hold 128 sequences at once.

### FILL 'ER UP

All the sequencers make use of the powerful Fill feature, which can be used to control parameters such as pitch, Velocity, and tempo (see **Fig. 2**). Adjust the green sliders by dragging with the mouse to specify MIDI values from 0 through



**FIG. 3:** *SoftStep Pro* can draw fractals in real time and use data extracted from the image for various control purposes. Bitmapped graphics files can also be imported and used to control other functions.

127. You can also set values by checking the Keyboard option, which summons an onscreen music keyboard you use to select pitches.

To use the Fill function, click on the Fill button of the step sequencer you are working with. You can enter the data for each step manually or use Source commands to automatically create different data patterns. Available patterns are Random, Random-Walk, Repeat the First Value, and Count Up from the First Value. The Modify commands change the values in various ways, such as adding a value to all steps, sorting them in ascending order, removing duplicate values, and scattering the values to random positions. That's quite a bit to keep track of, and fortunately, tool tips appear when you move the mouse over each button.

Quantize functions force the notes to conform to selected scales and chords. Five working buffers are marked A through E, letting you try various options and compare them. For example, say you have a random sequence, and you want to try some variations without discarding the original. You could copy the sequence to all buffers using the Copy/Paste feature and then move to the B buffer and quantize to a major triad, move to the C buffer and quantize to a whole-tone scale, and so on. Available options are major, minor, pentatonic, and whole-tone scales as well as octaves, fifths, and major and minor triads. A Jitter function moves notes randomly one step at a time.

## RANDOM EXCESS

*SoftStep Pro* offers several modules that create either random numbers or numbers that follow probability distributions. Random sequences are useful in many settings, but if you don't want pure randomness, you can use the Probability module to make certain outcomes more likely than others according to a probability table you create. A simple example would be to emphasize the tonal center of your piece by creating a probability-distribution table that makes the tonic
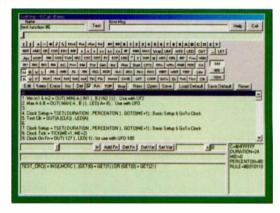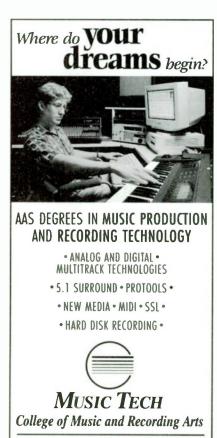


**FIG. 4: Users can create their own functions with the *User Function Editor*, a utility that offers an extensive toolkit for designing complex modules that can interact with the default resources the software provides.**

pitch most likely, emphasizes the intervals of a fifth and a third, and uses seconds and sevenths least often.

You can also get data from fractals or other graphic images. *SoftStep Pro*'s Fractal module draws fractal images on the screen in real time; you can use the patterns created by the changing positions of the pixels as they are drawn (see **Fig. 3**). The Clock module determines how quickly the drawing unfolds. The Image module works a little differently: import any image you wish (in BMP format), and *SoftStep Pro* will extract the color values at every *x-y* coordinate. You can use a Clock and a Counter to trace horizontally across the screen and then use the changing color values as loudness, pitch, or tempo information in other modules. The Image module includes a simple tool for drawing and editing fractal images to work with rather than importing them.

*SoftStep* can also read data from MIDI files, but they must first be imported and converted to *SoftStep*'s own MIDI-data format. You can create a *SoftStep* configuration that reads a MIDI sequence of a song, or even a tone row, and creates variations on it. You can choose to extract just the Velocity, Program Change, or Pitch Bend information from the MIDI file and use that data as a control source.

### GOING MICRO

*SoftStep*'s Tone module has extensive support for microtonality. A built-in selection of 103 tuning tables includes ancient Greek scales, just intonation, scales from various non-Western traditions, and scales developed by Harry Partch, Wendy Carlos, and Lou Harrison. If you want to create your own tuning tables, you can purchase a copy of the *MicroTone* editor (see the sidebar,

## MUSICAL WEATHER, SINGING DNA, AND EXOTIC TUNINGS

*SoftStep* has modules that let you use genetic information and other kinds of data in your musical materials as well as offer support for microtonal scales. If you want to go beyond the built-in features, you can buy three additional programs.

**DataBin** ($29) imports any text file and converts it to a binary format that *SoftStep* understands. That file could be a list of numbers, data organized in columns, or data delimited by commas. You could, for example, use a year's worth of temperature data to control changing tempos in a composition that depicts the passing of the seasons.

**BioEditor** ($39) lets you import genetic information (available on the Web and elsewhere) and use it in *SoftStep* configurations. *SoftStep*'s Bio-Sequencer module contains a number of built-in examples of genetic sequences, and the help files offer detailed information as well as links to additional sources. Those sequences produce not-quite-random

patterns that often translate into interesting musical results. *SoftStep Pro* limits you to a few built-in genetic sequences, however; if you want to add new information, you need *BioEditor*. (To learn more about work in that area by biologist Mary Anne Clark and *SoftStep* creator John Dunn, go to http://mitpress.mit.edu/e-journals/Leonardo/isast/articles/lifemusic.html.)

**MicroTone** ($49) is a program you can use to create customized tuning tables. *SoftStep* comes with more than 100 tuning tables, but if you have a hankering to create your own, *MicroTone* is the way to go.

Each program is easy to learn and use, and all have well-conceived editing features appropriate to the type of data they handle. You can download the programs for free to try them, but you have to purchase a license key before you can save the data to files that *SoftStep* can use. You can buy all three programs in a bundle for $89.

# The world's best selling USB MIDI interface now comes with a few added features...

## Keys.

Introducing the USB Keystation Series from MIDIMAN.

"Musical Weather, Singing DNA, and Exotic Tunings").

You might be wondering how *SoftStep Pro* could offer microtonality when MIDI uses only the 12-tone tempered scale. That's easy: it uses Pitch Bend messages before each note. Because Pitch Bend is a Channel message, though, if you want more than one note to sound at a time, you have to put each note on a separate MIDI channel. If the notes are playing quickly, you will hear a Pitch Bend "swoop," especially when the difference from traditional tuning is great. Consequently, if you really want to explore microtonality, you'll probably need to work with a synthesizer that has good support for it. Even with that limitation, microtonal support is a welcome feature.

## DO YOUR MATH

*SoftStep Pro*'s math modules support standard features such as adding, subtracting, dividing, or multiplying values. The program also has modules that combine functions, such as multiplying two numbers and then dividing the result by a third, and a module that works with percentages.

Modulus arithmetic is supported, and because the Western musical scale is itself an example of modulus arithmetic, the feature is useful. For example, say you have a random sequence of pitches, and you want a chord to play whenever the note C sounds, no matter what octave it's in. If you performed modulus division by 12 on the sequence of pitches, all the Cs would give a result of 0. You could then use the number 0 as a trigger for the desired chord. Modulus arithmetic would also allow you to compress all MIDI notes into a single octave.

Logic modules come in two general categories: logical and bitwise. The output of the logic modules is true (127) or false (0). Logic modules are for functions such as synchronizing two clock signals, delaying clock signals by a specified amount, changing state at every clock pulse, and producing the opposite of the input at the output (a logical "not" function).

Bitwise logic modules let you work with each bit of a number. They have many uses; for example, you could cre-ate as many as seven tracks of rhythmic patterns using the Rhythm module (a clock function) and then employ the Mask logic module to select which of those tracks, in any combination, will be sent to the output. The logic modules give you many opportunities to control your configurations in various ways and to make the musical results interesting and varied.

## MAKE YOUR OWN
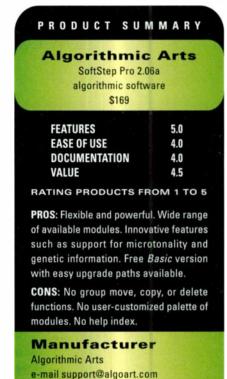
Although I just touched the surface, you can see that a module exists for just about anything imaginable. If you think of something the built-in modules can't easily handle, however, *Soft-Step* includes User Function modules for creating unique functions (see **Fig. 4**). Though they take a little effort to learn, there are some particular features, such as interactive graphics, that User Functions make possible.

For example, you can use User Functions to create a visual MIDI monitor that displays notes and their Velocities with vertical bars of varying sizes or colors. Or you might create a geometric design that dances on the screen in response to the music. Various drawing commands are available to create graphic images, and you can also import preexisting graphic files.

## IN A WORD

*SoftStep*'s help documentation is good: the tutorials are excellent, and handy tool tips appear when you move the mouse over different parts of each module. It would be useful, though, to be able to reach detailed help for each module directly from the Workspace. In addition, an index within the documentation would be helpful.

I would appreciate the ability to organize modules in my own way or create personalized palettes of modules that I use often. I also wish you could select several modules and then move, copy, or delete them as a group. As it is, you must deal with the somewhat disconcerting fact that you have to delete modules in different ways. You click on a box in the upper-left corner to delete most modules, but some require you to shift-click on little squares to delete

them; others even insist that you right-click on the modules themselves. Moreover, some modules disappear as soon as you delete them, but others ask you to confirm your decision with a dialog panel. Although I understand why those differences exist (you wouldn't want a Button module to be too easy to delete accidentally, for example), a group select-and-delete function would be a useful, timesaving feature.

Aside from those minor points, *SoftStep Pro* is a well-conceived program that's a pleasure to learn and use. It has many innovative and useful features; the bouncing-ball timer, microtonal support, and ability to use unusual data sources in your music are a few of my favorites. Customer support is excellent, and the upgrade paths let you start anywhere you want and dig in as deeply as your interests demand. Musicians interested in algorithmic techniques or interactive performance will find that *SoftStep Pro* delivers just about anything their musical imaginations can dream up.

*Peter Hamlin is a composer who teaches at St. Olaf College. He is also a member of the live electronic-music improv band Data Stream.*

# broadjam

# GRACE DESIGN

## MODEL 101

*Uncompromising sound from a classy yet affordable mic preamp.*

By Myles Boisen

To the observation that you can never be too rich or too good-looking, I would add that your recording signal path can never be too pristine. Microphones and mic preamps, after all, are the points at which the signal starts, so it only makes sense to ensure that each component is as clean, quiet, and accurate as possible (assuming that you're not intentionally seeking distortion, coloration, or the like).

The best gear comes at a price, and usually a steep one, which is why recordists working in personal and project studios must often settle for a compromise between quality and affordability. Fortunately, the gap between the two tends to narrow each year, thanks to low-end companies steadily improving their lot and to high-end outfits finding ways to reduce costs yet retain quality.

Grace Design, highly respected manufacturer of premium mic preamps such as the Model 201 and the Lunatec V2, is helping to further reduce the chasm between price and performance.

The company's new Model 101 offers the same dazzling specs and immaculate sound (or lack of sound) as its forebears, but it is priced at less than $700 (see **Fig. 1**). I used the 101 in numerous recording sessions during a period of several months; my only unanswered question is how in the world this small boutique audio manufacturer managed to put together a world-class preamp that's within the budget of nearly anyone with a mic and a part-time job.

## BUMPER TO BUMPER

The Model 101 is a single-channel, solid-state, half-rack mic preamp featuring a balanced, transformerless mic input and a high-impedance instrument input. The build is sturdy and elegant, and the unit's chrome front panel, which sports the same retro-modern metallic gleam and rounded contours as the company's Models 801 and 201, is so lovely that it practically begs to be smudged with fingerprints.

The 101's few front-panel features are laid out cleanly and labeled clearly. From left to right, it has a ¼-inch high-impedance direct injection (DI) jack (labeled HIZ), a 48V phantom-power switch, large easily grasped Gain and Trim knobs, a 75 Hz highpass-filter switch (labeled HPF), and a power switch beneath a recessed amber power-indicator light. Another recessed bulb in the top center of the panel glows green to indicate signal presence of −20 dBu and higher; when the signal level hits +16 dBu (a conservative 9 dB below clipping), the indicator turns red, bringing to mind the brake lights in the chrome bumper of a muscle car.

The Gain knob range is marked 10 dB (counterclockwise) to 60 dB (clockwise), with no intermediate calibration marks. Rather than use a standard linear potentiometer, Grace upped the ante by employing a precision stepped-gain switch with 11 positions. Each detent on the Gain knob corresponds to a 5 dB gain change, facilitating repeatable settings for anyone willing to count switch clicks. The continuously variable Trim control, useful for reducing gain in increments of fewer than 5 dB, is meant to be set to unity (fully clockwise in the 0 dB position) but can provide 10 dB of attenuation at its lowest (counterclockwise) setting.

The rear panel of the 101's sturdy all-metal chassis provides a balanced input and a balanced output on gold-plated Neutrik XLR connectors, along with a balanced ¼-inch TRS output. Power is supplied to the unit by a 6V wall-wart transformer. Although that may elicit grumbles, the alternative, a built-in toroidal AC transformer and linear power supply, would have resulted in a larger and more costly product. In addition, according to Grace Design, this is not your run-of-the-mill wall wart; rather, it is a high-quality power supply.

The lack of a polarity-reverse switch, a useful feature that Grace includes on its 801 and 201 models, is always vexing to me as a studio engineer. In multiple-mic setups or when interfacing with older gear in my studio, I often need to check and reverse polarity at the preamp. So when testing the Model 101, I kept a polarity-reversing XLR cable (one with pin 2 wired to pin 3 at the other end) within reach. However, considering the cost-to-quality factor of the unit, not having a polarity-reverse switch is an inconvenience I can easily overlook.

More troublesome is the absence of an indicator light accompanying the phantom-power switch. Because of the unit's snazzy front-panel sculpting, the position of the 101's 48V switch is difficult to see at a glance. Fortunately, that didn't lead to any mishaps during the test period; nevertheless, it's easy



FIG. 1: Although shy on features as compared with some competing boxes, the Grace Design Model 101 mic preamp and DI is remarkably transparent, coming about as close to the ideal of a "wire with gain" as you could ask.

## Model 101 Specifications

| | |
|---|---|
| Amplifier Type | solid-state, transimpedance, transformerless |
| Inputs | (1) balanced XLR; (1) balanced ¼" TRS |
| Outputs | (1) balanced XLR; (1) balanced ¼" TRS |
| Gain Range | 10–60 dB (mic); –10 dB–40 dB (line) |
| Maximum Output Level | +25 dBu |
| Output Trim Attenuator | 0–10 dB |
| Frequency Response | @ 40 dB gain, –0.05 dB:12 Hz–170 kHz (mic); 7.5 Hz–300 kHz (line) |
| Dynamic Range | 113 dB |
| Equivalent Input Noise | <–130 dB (50Ω source); <–128 dB (150Ω source); <–124 dB (600Ω source) |
| Total Harmonic Distortion + Noise | <0.00085% (@ 20 dB gain, +20 dBu out) |
| Signal-to-Noise Ratio | >92 dB |
| Peak LED Meter | –20 dBu (green); +16 dBu (red) |
| Highpass Filter | 75 Hz, 12 dB/octave |
| Dimensions | 0.5U × 6.5" (D) |
| Weight | 3.5 lb. |

to imagine someone accidentally disconnecting a condenser mic from the 101 while the phantom power is on, perhaps resulting in a loud pop and possibly even blown tweeters.

### HEAVYWEIGHT BOUT

The solid-state preamps in my studio are European and expensive, which means that I pitted the diminutive 101 against some heavyweight competition in recording sessions. Nonetheless, during the testing, the Grace preamp was predictably neutral in character and impressively close to the technical ideal of a sonically transparent "wire with gain."

For the 101's first outing, I paired it with a Baltic Latvian Universal Electronics (BLUE) Bottle tube microphone on singer Shelley Doty; the high-end characteristics of that mic-and-preamp combination highlighted an acoustic sibilance problem. Switching preamps (to a Focusrite Red 7) diminished the sibilance, and eventually I chose another mic to filter the vocalist's sizzle. The lesson is that accuracy (the 101's frequency response is virtually flat from 12 Hz to 170 kHz, according to Grace Design) is not always an advantage in the real world of recording.

As it turned out, that was the only occasion in which the 101 wasn't the right tool for the job. Indeed, I soon discovered that the same microphone in the same room sounded stunning through the 101 when used on a different singer. When set up for an impassioned male vocalist with a hint of

▼

**The highs extend into realms only dogs and test gear can appreciate.**

soulful grit, the signal chain displayed open highs extending into realms only dogs and test gear could appreciate.

A recorded A/B comparison showed that the Focusrite Red 7 had a warmer, chestier sound (a trait of transformers) with a bit more of an up-front attitude and more projection in the crucial 4 kHz range. The Grace, on the other hand, was beautifully airy without a hint of sibilance, and it conveyed more room detail along with a glossy realism. Although both preamps had merits, the next time the singer came into the studio, I set him up with his usual micro-

phone through the Model 101 and never looked back.

The 101 also garnered oohs and ahs for its performance on a track by poet Gilbert Marhoefer for his band, the Apes of God. I had already attempted Marhoefer's spoken-word piece on previous occasions, but until the 101 entered the equation, his voice never quite seemed to cut through the musical accompaniment. Matched again with the BLUE Bottle mic and followed by a Manley Variable Mu tube compressor, the Grace preamp worked like a charm to sharpen the vocal delivery without detracting from the tube warmth of its signal-processing companions.

Paired with an omnidirectional Neumann KM 183 as a distant acoustic-guitar mic, the 101 scored points for openness and radiant high harmonics as compared with a modified Neve 1272 preamp. The Neve seemed to concentrate and thicken the tone of the Taylor acoustic, emphasizing inner chord voicings. In contrast, the 101 brought low- and high-end details dramatically to the fore; discerned every pick stroke in a natural, realistic way; and made the guitar's top E string sparkle.

The 101 also seemed to optimize the omnidirectional characteristics of the KM 183—to the point that I picked up some distant machine noise coming from the other end of the building. Fortunately, the unit's low-cut filter effectively cut out the rumble without sacrificing the instrument's tone. For pop-music recording, the Neve sound will always have advantages and champions; for accuracy and presence, however, it lost that round to the Grace Model 101.

### NO-PAIN GAIN

The 101 has sufficient gain for distant-miking applications and can run at its highest gain setting without noise or artifacts. I teamed it with a Royer SF-1 ribbon mic on two rather delicate acoustic instruments: violin and Japanese koto. The Grace delivered adequate signal for +4 dBu recording to digital and analog media, and it ran clean right up to its +60 dB maximum. More important,

the 101's pristine signal path worked beautifully with the ribbon mic, and its ability to convey crisp detail was magical. (For recordists who require even more gain, Grace Design also offers a high-gain version of the Model 101, providing +70 dB maximum gain.)

On various members of the clarinet and saxophone families (recorded through a Lawson L47) and on trumpet (through the BLUE Bottle mic with a B2 capsule), the 101 lent a bright and incisive sheen to tube mics without sounding sterile, harsh, or buzzy. I also used the 101 on clarinet, marimba, violin, and bass drum for a session with composer Beth Custer featuring the Tin Hat Trio. A little reverb really made the instrumental tracks come to life and displayed vividly what the 101's clean electronics and extended high-end response can do.

### DIRECT FLIGHT
Used as a DI on a Fender Stratocaster guitar, the Model 101 sounded clearer and less noisy than the other preamp/DI boxes in my rack. Tonally, it came closest to my favorite guitar preamp, the Drawmer 1960, but the 101 provided cleaner, more articulate low notes and glassy, big-budget highs. There, with no EQ and no compression, was the crystalline direct sound, free of midrange muck, I had always imagined hearing from the Drawmer.

After that stunning revelation, I admit that I did not expect miracles from the 101 on direct bass. But it turned out to have more low-end depth and thump than most of my rack inhabitants, eclipsed only by the thunderous fundamentals of the Peavey VMP-2, my long-standing favorite for bass-DI duties. True to its flat-response character, the Grace sounded less thick in the crucial 150 to 300 Hz midbass octave than the other preamp/DIs tested, but don't call it thin just because it isn't fat. As was the case with guitar, the upper notes and harmonics were crisp, and the low end was wonderfully articulated and defined. I expect the 101 would really shine on a high-tech bass rig and allow the instrument to sit in a mix with minimal compression.

### GRACEFUL EXIT
For anyone accustomed to making a choice between premium and affordable gear, the Grace Design Model 101 is a real eye-opener. The solid, world-class microphone preamplifier can bring out all the subtleties heard at the microphone without imposing its own personality. In addition to that rare quality, the 101 provides high-end response that seems to extend without limit.

Although it may lack a few features common to other preamps, the modest single-channel unit outperforms any other preamp in its price range and offers serious competition even to legendary models costing three and four times as much. When you add it all up—uncompromising sonic purity, good looks, and affordable price—clearly, the Model 101 is an outstanding value for recordists working at all levels and in all styles of music. ●

## NEMESYS MUSIC TECHNOLOGY

**Christian and Lane Ultimate Timpani Library (GigaSampler)**

*By Zack Price*

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**T**he *Christian and Lane Ultimate Timpani* sample library of four CD-ROMs ($349) was produced and performed by professional percussionists Donnie Christian and Sean Lane. It contains just about every kind of timpani performance that you can imagine. The samples derive from a set of Hinger Touch-Tone timpani with calfskin heads.

EM ★★★★★
hot pick

You get six types of timpani performances: Hits, Hand Muffled Hits, Rolls, Crescendo and Decrescendo Rolls, Timpani Effects, and Timpani Extras. Each performance features playing with five types of mallets: cartwheels, which have thick, soft heads; three mallets with felt-covered heads, each with varying degrees of softness; and bare wooden mallets. The library also offers a set of timpani performances created by striking the timpani with fingers and hands. The combination of performances and mallets yields a complex matrix of sound choices.

### The Hits Keep Coming
The five mallet varieties each have six kinds of Hits performances: Hits, Hits Light, Pan, Pan Light, Bass Timpani, and Soprano Timpani. All Hits have eight Velocity levels, giving each hit a wide dynamic range. Hits, Hits Light, Pan, and Pan Light have separate left- and right-hand samples ranging from MIDI Note Numbers 36 to 57 (C to A) on the left to 60 to 81 (C to A, one octave higher) on the right. That keyboard layout makes it possible to play sustained crescendo, and decrescendo rolls in real time. Hits Light performances simulate the effect of a light cloth placed on the timpani head, which is useful for creating articulate passages. The patches' hit and fade levels can be controlled with the Mod wheel. Pan and Pan Light performances map the timpani pitches

from right to left as you move up the scale.

Ranging from C36 to A57, the Bass and Soprano Timpani hit patches offer nice special effects. The Bass Timpani is a standard timpani pitch-shifted down one octave. You can use the Mod wheel to control the samples' attack and release times. That can provide some cool, eerie effects, especially if you use some of the effects processors available in *GigaStudio*. The Soprano Timpani is a normal timpani sound, but it is pitch-shifted up one octave. It has an interesting, almost Oriental-sounding tone; I used it to play a Chinese-inspired melody, and it sounded great.
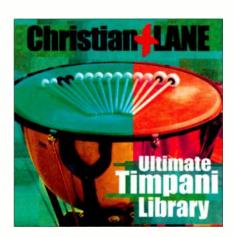
Each time I played A45 on the Green Soprano Timpani patch with a hard velocity, the sound erroneously switched to a regular timpani instead of soprano sound. However, you can go to Christian and Lane's Web site (www.dssoundware.com) to download an articulation file that corrects the problem.

The Hand Muffled performances contain single hits that are muted with the hand after the initial stroke. The Hand Muffled performances are arranged similarly to the Hits performances, with the following exceptions: there are no hand-muffled Bass or Soprano Timpani patches, and there are no separate left- and right-hand hits. That's as it should be—in performance, one hand would be busy muffling the head.

### Merrily We Roll Along
Playing realistic rolls is possible using the Hits patches; however, doing so uses considerable polyphony. The Rolls patches reduce polyphony demands and generate smooth rolls. Each note (C36 to A57) in the Rolls patches (Rolls, Rolls Light, Rolls Pan, and Rolls Pan Light) is mapped at four Velocity levels. Furthermore, rolls in the softer dynamics of the lower pitches are slower than rolls of the higher-pitched notes, which is how they occur when performed live. Each note has a release trigger that allows for a realistic release when you need a roll shorter than the actual sample. Considering that each Rolls note lasts about 30 seconds, you often will need shorter releases. You also can control the release time with the Mod wheel.
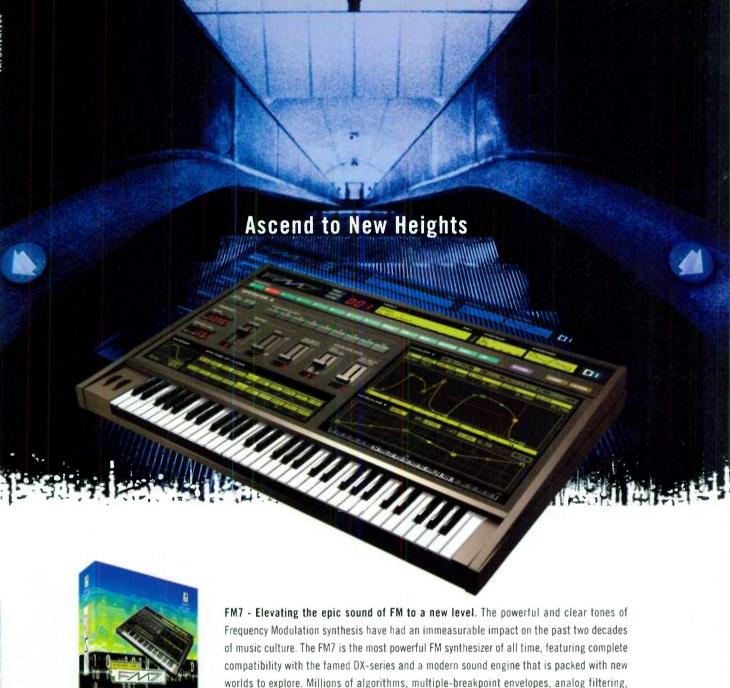
The Crescendo and Decrescendo patches let the user generate smooth, realistic crescendo and decrescendo rolls with a minimum of polyphony. As in the Rolls patches, notes are mapped from C36 to A57. The speed of each roll (short, medium, and long) and its dynamic direction are determined by playing notes C60 to A69 before playing the desired roll note. Again, the Mod wheel can control release times.

The Timpani Extras patch includes glissandi, bowl hits, and my personal favorite, a glissando with a cymbal on the head. It's an eerie sound, but I like it. The Fingers and Hands performances are similar in many ways to the Hits. Both have Hits, Hits Light, Pan, Pan Light, and Bass patches, and each corresponding patch is laid out in the same way on the keyboard. However, the Fingers and Hands performances have only four Velocity levels, ranging from a tap with a single finger to a fortissimo hit with the entire hand. Additionally, the Alto Fingers and Hand patch is pitched lower than the Soprano Hits patches. Even though probably no music is written for those patches, they are fun to play.

### Decrescendo
Most of the performance parameters are pretty straightforward. However, more specific information concerning mapping and key switching would be nice. Unfortunately, the only documentation provided is a Microsoft *Word* file that duplicates the information in the CD-ROM booklet. Nonetheless, the Christian and Lane Web site offers additional information and tips about creating realistic performances.

Despite that and the easily corrected

*The **Christian and Lane Ultimate Timpani Library** from NemeSys Music Technology provides an impressive collection of realistic, expressive timpani sounds and performances.*

patch glitch mentioned previously, Christian and Lane have labored through a Herculean project to provide the best timpani sample library you could hope for. If you need to access a variety of timpani sounds on a regular basis, get the *Ultimate Timpani Library*. It truly lives up to its name.

**Overall EM Rating (1 through 5): 5**

NemeSys Music Technology; tel. (512) 219-9181; e-mail sales@nemesysmusic.com; Web www.nemesysmusic.com

# F7 SOUND AND VISION

## Concept:FX3

*By Jeff Obee*

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**M**ichael Oster is the sole guiding force behind F7 Sound and Vision and the producer of two high-quality CD-ROMs of sound effects in the Concept:FX series. *Concept:FX3* ($89.95) is Oster's third CD-ROM of 16-bit, 44.1 kHz AIFF and WAV files. The collection is aimed at video and film post-production, radio and TV broadcast, multimedia and game development, and Web sound design. All three CDs are available in the Concept:FX Pro Bundle for $169.95.

The sounds come from a cornucopia of sources, and they tend toward the experimental and the unusual. Sound sources include Oster's voice, old tube radios, microcassette recorders, a ¼-inch tape machine, microphones from both ends of the quality and price spectrum, office supplies and paper products, an antique movie camera, and old clothing. Oster also derived sounds from a Roland JP-8000, a Sequential Circuits Six-Trak, an ARP Odyssey, and E-mu modular synths as well as old effects pedals, guitar, and bass. A Korg Kaoss pad further assisted with sound manipulation.

Oster fed microphone and line sources through a 2-channel Amek System 9098 Dual Mic Amp and direct injection box and Apogee AD-8000 and Rosetta digital converters to either an HHB CDR850 CD recorder or Pro Tools 24/Mix system. Once in Pro Tools, the samples were sliced, diced, and processed with plug-ins. Oster deliberately refuses to use presets or take notes, ensuring that he won't fall back on redundant production methods.

**Conceptual Menu**

The CD-ROM contains folders of AIFF and WAV files, each with seven identical subfolders that are organized according to the basic characteristics of the sounds they contain: Bites, Comic, Grunge, Hits, Longer, Small, and Static.

Oster's sound palette serves as a springboard for today's sound-design production work. A random sampling of the 282 sounds in the Bites folder revealed fluttering flybys; panning resonant noise bursts; electronic animal roars; alien sci-fi snippets; quick, chirpy stings; machine noises; hollow-sounding whooshes; crispy crackles; and scorching sizzles. All in all, the versatile panorama of sound design ranged in size from 30 to 770K.

*Concept:FX3*, a sample CD-ROM from F7 Sound and Vision, has a generous variety of special effects that can be used for video and film post-production, Web sound design, multimedia and game development, and more.

The files stand well on their own. However, you can easily append, crossfade, convolve, and mix them into a longer, altogether different sound. I chose BT002 and BT003, blended and reversed them, played with some pitch shifting, and ended up with a sound very different from the originals.

*Concept:FX3*'s Comic sounds lack the humor of those in *Concept:FX2*. Appending a few and sculpting them further can provide amusing results, though, and the sonorities of the sounds work well together. The Static folder contains distorted buzzes, noise, and, yes, all manner of searing, electrical static. The Rip files in that folder are exactly the tearing, shredding noises that one would expect.

The Hits folder brims with brief barks, smacks, wails, thuds, and squeals. It holds some unusual snare sounds—not the typical Oster offering, but a nice addition. The Grunge folder outdoes them all in terms of rude, intense samples: you get loud, raw, machinelike bits; piercing distortion; deep-pitched rumbling; and much more. Brace yourself.

The Longer folder contains some beauties. Ambnce01 gives the impression of an airport ambience fed through a transistor radio, whereas Ambnce03 has a droning, netherworld quality. Bgds03 is a slightly resonant dull-noise sample that churns and chugs in a consistent rhythm of approximately 127 bpm and is excellent for looping. Voxsim02 has the spiky rhythm of a speaking voice, blurred by just the right amount of distortion. Many files contain loop markers.
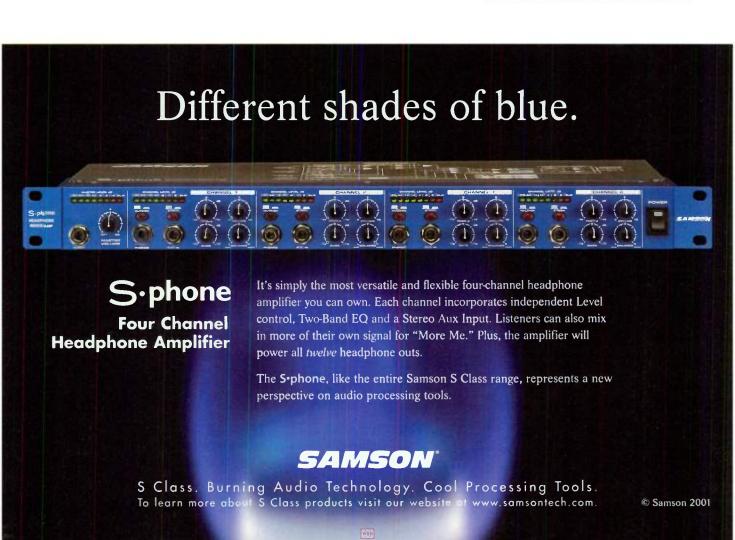
### How It FX Me

F7 Sound and Vision has released another fine collection of creative sound designs. The zany file names of the earlier offerings in the series are replaced by a more pragmatic naming system, which makes things less entertaining but easier to grasp in organizational terms. In contrast to *Concept:FX2*'s 305 medium to long samples, *Concept:FX3* gives you 1,292 short to medium samples. The shorter file lengths make them excellent base material for manipulating to your needs. The price is right too. I recommend it.

**Overall EM Rating (1 through 5): 4**

F7 Sound and Vision; tel. (813) 991-4117; e-mail f7sound@gte.net; Web www.f7sound.com

## PROSOUND COMMUNICATIONS
### Robotalk
*By Steve Broderson*

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**D**espite the popularity of do-everything multi-effects processors, stompboxes have held their ground in the hearts and rigs of many musicians. A patchwork pedalboard of unusual effects is the mark of individuality, and to some players, the more obscure the effect, the better. Enter Robotalk ($280), a filter pedal that combines three effects in one device.

Robotalk gives you a wah-wah, an envelope filter, and a sample-and-hold effect in a hefty bare-metal box. Each pedal is hand-built with U.S.-made parts. Robotalk has two footswitches: a Function switch, which toggles between the envelope filter and sample-and-hold effect, and an Effect On/Off switch, which is not a true bypass switch. An LED indicates when the effect is on.

The Robotalk offers three ¼-inch jacks: an audio input, an output, and an expression-pedal input. You activate the wah-wah effect by plugging a voltage-control expression pedal in to the expression-pedal jack. When an expression pedal is connected to Robotalk, it overrides the Function switch and locks the unit into Wah-Wah mode.

Most standard expression pedals work with Robotalk; I used Roland's EV-5, which the manufacturer recommends. Two 9V batteries power Robotalk, but it has no provision for an AC adapter.

### The Envelope, Please
As a standard envelope filter, Robotalk works well and sounds great. The filter characteristics are controlled by two tiny trim pots labeled Range and Freq. Range adjusts the amount of gain required to make the filter kick in, and Freq adjusts the EQ curve. Adjusting the trim pots is tedious and difficult because of their size and lack of markings, and identifying the settings is difficult.

The two other controls, Volume and Rate, are larger and clearly marked. The Volume dial drives the pedal's overall output, and the Rate dial directs the speed of the filter's sample-and-hold portion.

In Sample-and-Hold mode (which is misleadingly termed Arrpegiator *[sic]* by the manufacturer), the filter setting changes randomly at a speed that's determined by the Rate control. The speed ranges from a couple of seconds per step to a fast warble. At higher speeds, the result is a rhythmic burbling reminiscent of an analog synthesizer. The Range control doesn't function in Sample-and-Hold mode because the filter sweep no longer depends on your input level. You can set the sample-and-hold frequencies using the Freq control.

Robotalk's filter sweeps in Wah-Wah mode are smooth. The wah-wah's range seemed to be wider than that of Dunlop's familiar Crybaby pedal. After using Robotalk for a while, I wished for a way to use all three effects modes at one time without disconnecting the expression pedal.

### Way Out
The Robotalk pedal's only drawbacks are the small trim pots and the boutique-level price tag. Without the sample-and-hold circuit, Robotalk is merely a pricey envelope filter and wah-wah pedal. Still, Robotalk is a solid stompbox that drips with custom-built retro chic. Collectors of unusual stompboxes will want to make a space on their pedalboard for Robotalk.

**Overall EM Rating (1 through 5): 3.5**
Prosound Communications Inc.; tel. (818) 996-8603; e-mail info@prosoundcommunications .com; Web www.prosoundcommunications.com

## MIT PRESS
### Machine Musicianship
*By Douglas Geers*

● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ● ●

**M**usicianship is the set of skills most musicians need to understand and perform music well, including the ability to recognize and repeat a phrase of melody, the ability to find a tune's meter, and the ability to identify a chord from the notes that are playing. Luckily, musicians are born with some of those skills and can learn others through experience. Still, musicianship often requires years of practice and performance to achieve a sophisticated level of mastery.

What about computer musicianship? Computers are used all the time to play, record, and edit music, but can they be programmed to hear and understand music as people do? If so, how? Those questions and others are answered in a new book titled *Machine Musicianship* ($47.95), by Robert Rowe, a composer and professor of music at New York University.
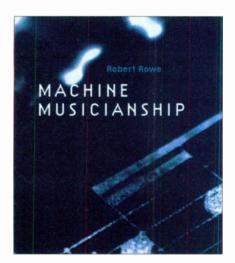
Rowe's book is written for the DIY music hacker. Accompanied by a CD-ROM of software examples and utilities, the book takes users through ten chapters of progressively more sophisticated machine-musicianship topics, ranging from general explanations to discourses about algorithmic composition, computer improvisation, and musically responsive Web sites. Rowe explains concepts and then describes how to program computers to accomplish the tasks discussed.

### Chapter and Verse
After chapter 1's introductory discussion, Rowe begins the real work in chapter 2. There he explains how to write the C++ code that instructs a computer to name chords as they are played on a MIDI keyboard, from simple triads to thick jazz chords and inversions that can be named only by examining their musical context. As he proceeds, Rowe explains his algorithms and the theories behind them, which is informative but at times complex and digressive.

Chapter 3 covers *beat tracking,* a technique in which a computer finds and follows the beat within an incoming audio stream—something easy for people but quite tricky for a computer, Rowe explains. He then takes the reader through his own beat-tracking algorithm, a fascinating method in which the computer makes multiple simultaneous guesses of the tempo, judges them, and then crowns one as the winner. It works well and sounds useful for solo performances.

With readers and their computers able to analyze pitch and rhythmic data, Rowe zooms out to larger issues. In chapter 4,

Prosound Communications' Robotalk combines an envelope filter, a sample-and-hold effect, and a wah-wah in one stompbox.

Robert Rowe's *Machine Musicianship* shows you how to supply a computer with the many skills needed to create and analyze music.

he succinctly discusses how computers can recognize musical motives and other patterns. Chapter 5 examines useful compositional techniques for real-time performance, such as cueing sequences, transforming input, and varying stored musical materials. Chapter 6 covers systems that learn over a period of time, and the final chapters investigate interactive improvisation, interactive multimedia, installations, and possibilities for the future.

## Extra Goodies

The book's CD-ROM has the code needed to realize those computer-musicianship skills and, as a whole, is a well-organized gold mine of useful tools for interested musicians with computer-programming chops. The programs, written in C++ and Cycling '74's *Max*, give the user tools for accomplishing just about every basic interpretive function she or he might want a computer to perform upon receiving musical input, from chord identification to genetic algorithms. Rowe includes the source code and executables for those computer routines, so even those interested in the ideas but not the programming can bring to life what is discussed in the text.

Unfortunately for PC users, the applications on the CD-ROM were created for Macs, and the C++ code was optimized for the Mac as well. But the CD does contain 18 tracks of audio examples and several QuickTime movies that are playable on Mac and PC.
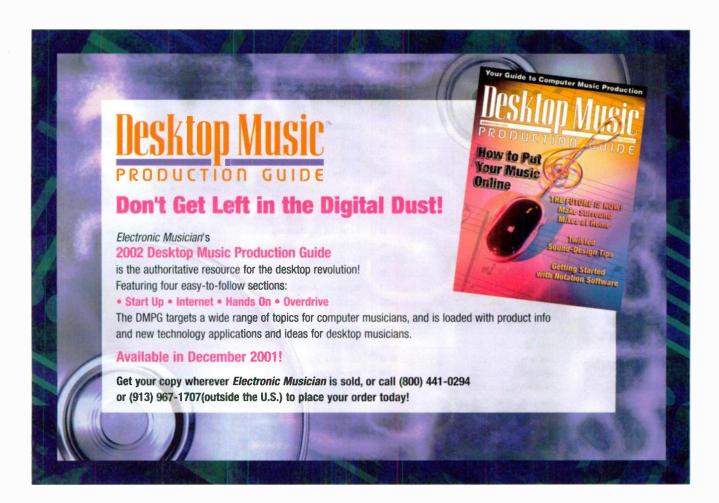
Rowe's writing style is generally clear, but he keeps up a fairly brisk pace and assumes that the reader is an educated musician. For those who aren't versed in music theory, things will get confusing rather quickly. In chapter 2, for example, Rowe discusses pitch-class equivalencies and "normal form" in relation to recognizing and classifying triads. But if you're brave enough to plow through it, *Machine Musicianship* could help you turn your old computer into your new bandmate—able to do anything but help you carry your gear to the next gig. ☻

**Overall EM Rating (1 through 5): 4**

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# Box of Rain

There are times when a person may experience something truly and literally incomprehensible—that is, something the meaning of which the mind cannot fully grasp. Different experiences may evoke various feelings, but incomprehensibility is a glaze that dissolves focus even while you may be single-mindedly riveted by what is occurring. When everything you know is wrong, you feel separated from any sense of understanding the world, and that throws you for a loop. All your regular points of reference are washed over by waves of confusion and shock until they are worn down to a rubble of questions about things that no longer fit neatly together as they once seemed to. Mind boggling, world shattering: overhyped colloquialisms that today describe an indigestible daily reality.

Now multiply that effect by hundreds of millions, even billions. It is nothing less than a global wrench in consciousness.

How can a planet thrown so off-kilter begin to search for balance? There are many good answers, but I believe one of the best is music. Music speaks internationally and can be eloquent in any language. Music expresses feelings and thoughts in ways that words alone cannot. Music reaches into the heart to touch the spirit, and that is where healing begins.

Clear Channel Communications' idea of issuing a list of songs its radio stations should not play in the wake of the September 11 attacks may have been presented as a desire to avoid further hurt, but what is needed is exactly the opposite. Perhaps certain music will upset some people, and it is even possible that music could provoke people to bad actions, but the power of music to draw people together and remind them of the miracle of humanity is far greater and more important. It is not worth risking the baby by throwing out the bathwater.

A Bach sonata, Coltrane's *A Love Supreme,* Ravi Shankar spinning a flowing raga, a silky samba, the ancient voices and drums of the Middle East or Africa or an anguished Chicago blues—all can transport people to a place where themes of life are set forth, embellished, and woven into a fabric in which each of us is but a thread, and those of us no longer here leave holes that may appear small from the viewpoint of the entire fabric but are huge and gaping from the perspective of the strands around them.

Music helps people relate that which is clearly beyond them to that which they see, do, and feel every day. This connection, where the cosmic, cerebral, physical, and emotional converge, is a genuine meeting of the spirit, and it is this process—and the awareness that it is happening—that can help us begin to regain our balance. Although the old mix of elements has been destroyed, a new mix will be created.

Whether you are a player, a listener, or a dancer, this is a good time to listen to music that touches you. Find solace in favorite music you know and discover new perspectives in music you have not heard before. If you make music, let it take you where it wants to go. Sit down and lose yourself in it or let it play behind whatever else you're doing. Don't be afraid to understand other people; you can do it by listening to their music.

Regaining balance is only the start of what must happen in the world from this point, but it is a crucial first step. I make no claim to know the Right Thing to Do, but I know that any hope of wisdom for charting our course from today will be strengthened by balance. The greatest global coalition will happen as different people around the world become aware that they can develop a bond. Fear is a powerful motivator for cooperation, but understanding fosters joining together, and that is a stronger force.

I can say nothing in the six or seven hundred words of this column that could make any real difference, but almost anything that could make a difference can be said in a few well-played notes. They may not explain the incomprehensible, but they can provide a picture of life that includes the incomprehensible, and that's a beginning. ◉

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