A PRIMEDIA Publication

Electronic Musician ®

January 2002

THE YEAR'S 33 HOTTEST PRODUCTS

HIT-MAKING SECRETS
Five top producers reveal all

'2002 EDITORS' CHOICE

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WRH

NEW! 24-TRACK 24-BIT HARD

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MACKIE. 24TRACK/24BIT DIGITAL AUDIO HARD DISK RECORDER   **MDR 2**

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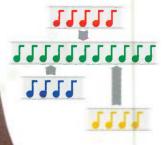
LINEAR vs NON-LINEAR RECORDING.

Tape-based recorders (and even some hard disk recorders), record in a linear manner. They record data in a continuous "stream" (the green stream of notes in our drawing).

If you want to change something on a track, the recorder permanently erases the old data and records new data over it (red notes in the drawing below).

The non-linear MDR24/96 records any number of alternate segments on separate parts of the hard disk and automatically "punches them in" during playback. You never lose your any previous work.

DISK RECORDING FOR $1999*
linear MDR24/96 at a special intro price.

Greg Mackie's goal has always been to deliver professional creative tools at reasonable prices.

The new MDR24/96 is our price-busting commitment to nonlinear hard disk recording. Read about its rich feature-set and you'll see we didn't skimp anywhere.

PLUG AND PLAY RIGHT OUT OF THE BOX.

Now you can retire those tape-based digital recorders and enjoy the convenience and enhanced creativity that comes with instant-access hard disk recording.

Ready to go with twenty-four

channels of analog I/O *already installed*, the MDR24/96's familiar tape recorder style controls and simple operation will have you tracking and mixing down in no time.

TWO REMOTES. TWO REMOVABLE MEDIA OPTIONS.

Our new recorder uses the same affordable, removable recording and backup drives as our landmark HDR24/96 Recorder/Editor. Mackie Media Project M90 pull-out drives and 2.2GB ORB™ disks make saving and storing projects a breeze. (And of course the MDR24/96 ships with a 20Gb internal hard disk to get you started).

It also uses the same compact Remote 24 and ultra-mondo Remote 48 controllers.

Removable ORB™ drive
(*Songus Backupus Sneakernetai*)

NON-LINEAR MEANS NEVER HAVING TO SAY YOU'RE SORRY.

The MDR24/96 is a *non-linear, non-destructive* recorder. That means you can punch-in "over" a section of a song as many times as you want without erasing it — and then choose the take you like best later (see explanation on the lefthand page). Non-linear recorders also use drive space more efficiently than linear or "tape mode" hard disk recorders.

You can do basic cut, copy and paste editing

Remote 24
(*Getzajobdonus Economicai*)

with 999 Un-Do levels using the MDR24/96's front panel controls.

MIX AND MATCH 'EM!

Run MDR24/96s alone or in pairs with any analog or digital mixer. Slave it to the HDR24/96 Recorder/Editor and Remote 48 controller and scrub tracks right down to the waveform level for ultra-precise editing. Or install MDR24/96's in your 'B' and 'C' rooms and edit projects on an HDR24/96 in the 'A' room.

Remote 48
(*Remotus Deluxai Way-coolus*)

*GOTCHA-FREE FINEPRINT!

No, you don't have to buy six MDR24/96s to get this price. But we *do* have to point out that $1999 is suggested U.S. street price and that your price may vary. This is a limited time, introductory price and it *will* go up after our bean counters come to their senses and make us stop practically giving MDR24/96s away.

So don't tary, Harry. Don't wait, Kate.

Check out the no-compromise MDR24/96 24-track hard disk recorder at your Mackie dealer today. ■

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I N S I

FEATURES

Cover by Dmitry Panich

DEPARTMENTS

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116

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90

Trivial Pursuits

When so many have recently been killed, maimed, widowed, orphaned, or impoverished, we would do well to reflect on what we do each day and consider its importance. Certainly we aren't all cut out to be heroes, pulling survivors out of burning buildings, catching criminals, or defending the nation. But with that understood, it's good to seek perspective, especially when we are warm and dry (though perhaps no longer feeling as safe as we once did) in our studios and offices.

Having given blood, donated money, participated in public discussions, or volunteered for a nonprofit cause (September 11–related or not), at the end of the day you still have to make a living. If you are fortunate enough to still be employed making widgets, hopefully somebody still needs those widgets. And you still have to eat, so widget making is vital from that viewpoint, even if it seems trivial in the scheme of things.

Fortunately, what we musicians and producers do is not quite as trivial as it might seem at first glance. While producing music certainly isn't in the same league with fire fighting, it is good to realize that some of the firefighters probably look forward to listening to music when they finally return home or to base for much-needed rest and relaxation. Their families—and the families of those less fortunate—may turn to music for encouragement or consolation. Volunteer organizations often use music to help focus their message and inspire their supporters. Music helps us communicate in a powerful way at a time when people especially need to connect with each other. So although we musicians don't work in a vital industry in the sense that, say, medical staff do, we may be contributing more than we imagine.

I thought about that many times while we were putting together this year's Editors' Choice Awards. On the face of it, coming to the office to edit a magazine seems less inspiring than helping those in need. But our readers rely on us to deliver specialized information and advice that helps them make the music of their dreams. Helping people make their dreams come true is a pretty fine way to spend one's days, even in wartime; looked at that way, maybe creating this magazine is a valuable contribution after all. And if those musical dreams in turn help those who listen to the music, we've contributed in a small way.

So even while we contemplated the latest national disaster and struggled to regain our balance, we **EM** editors rededicated ourselves to the task at hand: making this a better magazine and making the tenth annual Editors' Choice Awards the best yet. Once we got into it, we realized that some of these products were downright amazing and could help musicians create inspired music—a Very Good Thing, especially in these grim times. And that led us back to remembering that our job is really to support your musical efforts.

It is our sincere hope that this issue and every issue will turn you on to a product or technique or concept that helps you produce a better song, a song that will make a difference to someone who needs to hear it. It may be a small contribution to making this a better world, and it isn't heroic, but it's worthwhile nevertheless.

www.emusician.com

STORY #23
Chuck Johns   Queens, NY

GOOD KARMA. WICKED SOUND.

KORG

KARMA ™ (Kay Algorithmic Realtime Music Architecture) is a trademark and patented technology licensed from Stephen Kay, Karma Lab LLC. www.karma-lab.com

THE NEW DIGITAL WORKSTATION
FOR PEOPLE WITH MORE SENSE THAN DOLLARS.

If this all-in-one digital workstation looks familiar, that's
because the new Yamaha **AW2816** strongly resembles its TEC Award-
nominated big brother, the AW4416. In terms of features and performance they're remarkably similar. So, the AW2816's price –
just $1,999* complete – makes excellent financial sense. Once again, Yamaha gives you more for less.

- 16+2 track, 24-bit recording with no data compression (44.1 or 48Khz)
- 28 Input channels (including digital stereo input and dual stereo effects returns)
- Dynamic automation of all mix settings, including fader movements, parameter changes and scenes
- 2 Assignable 32-bit effects processors
- Premium Yamaha CD recorder for data and audio

- I/O expansion slot for digital interface options or the new WAVES® Y56K processor card
- Large, back-lit 320x640 display screen
- MIDI control of computer based recording systems
- 4-Band parametric EQ and dynamics on all channels, all the time**
- 18 Busses, plus comprehensive digital patch bay, with easy routing

⊛ YAMAHA

CREATING 'KANDO' TOGETHER

www.AW2816.com

MAE LAROBIS

I WANT MONEY

I realize "Working Musician: Follow the Money" (November 2001) was created with the novice in mind. But some salient facts weren't included.

The authors state that the record company recoups its expenses for marketing the album. It would also be helpful to indicate which items are not recoupable. For example, music videos are often 50 percent recoupable.

The article provided a good idea of the benefit of songwriting but didn't make it clear that the mechanical royalties are paid before the album recoups. Therefore, the songwriters can earn significant money even though the recording artists are still in the red.

Artist royalties are generally paid on 75 to 80 percent of units to allow for discounts, premiums, promotional giveaways, and so forth. That reduces the recoupment schedule and increases the break-even number of units. Many artists are surprised to learn that they receive royalties on only 75 percent of units sold.

Producer royalties are often on a different recoupment schedule and don't come out of the artist royalties. In that case, it is advantageous to have producer points and songwriter credits.

Steve Corn
via e-mail

I read "Working Musician: Follow the Money" with interest. I would love to see more data and discussion, and I highly recommend Courtney Love's diatribe (www.salon.com/tech/feature/2000/06/14/love) for an alternative view. In addition, what about music technology? The economics that drive research, development, and production of music technology determine, to a large extent, what technologies musicians can use. It's a complex picture, and I would like to understand it better.

Returning to the article, the hypothetical example seems to have a mistake in math. The authors state that the record company recovers its investment at a rate of $5.75 per unit sold, but in fact, the investment is also paid off by the artist at $2.25 per unit. Therefore, the record company recovers its investment after 62,500 units (ignoring the producer royalty), not 86,957. In the example, at least half of the advance is marketing money controlled by the company and never paid to the artist. Because the company benefits most from marketing, shouldn't it pay its fair share? Why doesn't the producer share those costs as well?

Another test of fairness is who ends up with the money. It would be interesting to see the total income distribution to artists versus record companies. If record companies are really taking a risk, analysts should be able to find large variations in profit between companies and between years, attributable to whether the companies picked the right artists. Also, recording companies that take risks should sometimes fail. Is there any data along those lines that measures risk?

Roger Dannenberg
via e-mail

Authors Eric Leach and William Henslee reply: Roger—Your letter brings up several good points. Courtney Love's "diatribe" is quite interesting. An often overlooked fact concerning contracts is that they are completely voluntary. We recommend to any artist that if you can find a better deal regarding worldwide promotion, manufacturing, sales, and distribution of your music, by all means, take it. The problem is that new artists have almost no bargaining power. Established artists can, and do, negotiate much more favorable terms in their contracts.

To clarify our math "mistake": the issue on point is the number of records, from a very simplified hypothetical standpoint, that a record company must sell to recoup its advance purely out of its profits. Technically, artist royalties are an expense, which actually reduces the profits rather than increases them. You are correct in your assertion that artist royalties are retained instead of distributed and subsequently recovered. Similarly, marketing arrangements can vary significantly from contract to contract; our hypothetical contract was meant only for broad reference.
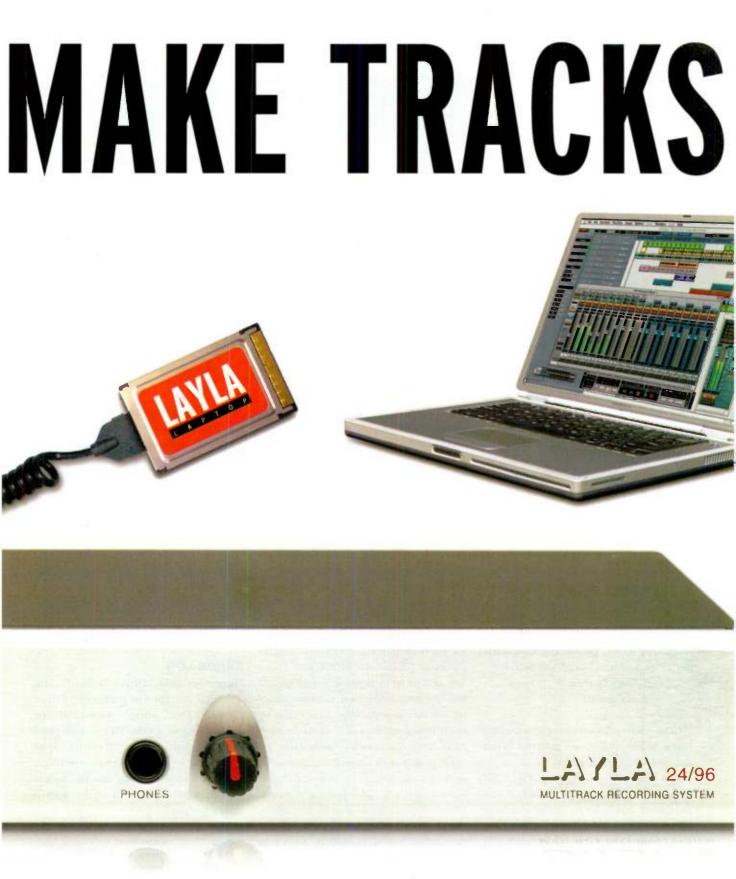
Your "test of fairness" point is well taken. Although such hard data is not at hand, in the past several years, the failure or buyout of dozens of smaller record companies suggests that only large independently successful companies have the resources to bear such risks on a global scale and profit accordingly. It is true that, in any business, a company that has financial resources to draw from is much more likely to weather the profit-loss fluctuations of high-risk ventures.

TO USB OR NOT TO USB?

Reading your review of the Tascam US-428 (September 2001) reminded me of the glaring failure of EM and most other related industry publications to blow the whistle on the vain attempt of the USB standard to properly support MIDI. I recently read a column in another magazine that stated:

1. USB MIDI interfaces suffer from excessive jitter (timing "slop");

2. the USB specification is at fault for that (the USB people did not talk to the MIDI people until some critical parts of

MAKE TRACKS

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

the USB specification were already set in stone; therefore, timing issues could not be properly addressed);

3. the few devices that supposedly solved the problem used solutions that were applicable only to MIDI software from the same manufacturer;

4. older interface technologies (serial port, parallel port, ISA card, internal sound hardware, and so on) have much better timing stability.

I know of at least one USB device that can also be used with the old serial port instead. Used with the serial port, timing is okay.

Silence fell across the pro-audio publishing community in the wake of the column. Every maker of MIDI interfaces seemed to hop on the USB bandwagon. Clearly, a lot of money is at stake. Those manufacturers advertise, and I suspect some strong-arming is going on.

Like other USB devices with MIDI capability, the US-428 cannot be relied on to provide the MIDI timing stability required for professional use. I am sure that in all other respects it is a quality product, but the review should have warned readers about using the unit's MIDI ports if tight performance timing is important. I am less concerned with taking Tascam to task than with the failure of the publishing community to look out for the interests of its readers.

USB is okay for some things (audio, mice, keyboards) but isn't suitable for MIDI. The sooner people give USB MIDI hardware a decent burial, the better!

Cary Cornett
via e-mail

Cary—Your letter raises important issues that need clarification, so I contacted several industry leaders for comment.

Tom White, president and CEO of the MIDI Manufacturers Association (MMA; the organization responsible for setting standards related to MIDI), stated that "the silence Cary Cornett hears is not due to a conspiracy as much as to the fact that the industry does not have consensus on how to use USB for MIDI. There is more than one way to do MIDI over USB, and some methods are arguably better for timing accuracy than others." White explained that the method designated by the USB Implementers Forum (USB-IF) was evaluated

by the MMA, and "we found that the level of jitter varied considerably among systems with different configurations, making it difficult to make any absolute claims." The MMA therefore declined to support the USB-IF MIDI Specification as currently written, but it "does not agree that USB MIDI does not work."

It's important to remember that companies making USB MIDI devices often don't follow the USB-IF MIDI Specification and instead use time stamping and other methods to provide timing accuracy as good as or better than what is available through serial devices.

Jim Cooper at MOTU agreed with White's assessment, stating that trashing USB MIDI as a whole is like throwing the baby out with the bathwater. He said that MOTU took a hard look at USB before jumping in and decided that "for truly professional use, time stamping was the way to go, allowing us to achieve MIDI timing accuracy of 0.33 ms for every MIDI event—far better than serial MIDI."

The quality of the device drivers is another important variable. As the technical staff at Tascam explained, "Most people who criticize USB as an audio tool do so on the presumption that the device's drivers are written according to the standard protocol, in which the device polls the USB bus on a regular interval, usually once every millisecond." That may be fine for most computer peripherals, but audio devices are much more demanding. Tascam is quick to point out that "the US-428's drivers were created and specifically optimized for audio and MIDI performance" and greatly exceed the

specifications for typical computer peripherals. Tascam states, "A well-written driver and a computer optimized for audio will much more heavily influence MIDI timing as opposed to the USB spec itself."

In short, USB has the potential to deliver far greater performance than the old serial interfaces when used properly. Proprietary hardware and software combinations offer less flexibility than generic serial interface systems, but many users are more than willing to sacrifice some flexibility to achieve much greater performance and, in some cases, the cross-platform capabilities and convenience provided by USB.—David Rubin

ERROR LOG

November 2001, "Direct Action," p. 48: The first sentence in paragraph two of "Ins and Outs" should have read, "In some designs, a tube DI's unbalanced input is multed directly off its input jack—that is, it's wired in parallel with the input so what goes into the box is exactly what comes out at the unbalanced output."

The beast in a titanium cage.

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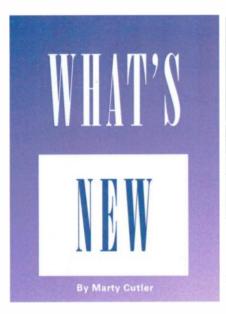
76 KEYS · OVER 1000 SOUNDS · D-BEAM CONTROLLER · SRX-EXPANSION ·
FLEXIBLE LINEAR AND PATTERN RECORDING · REALTIME PHRASE SEQUENCING ·

WHAT'S NEW

By Marty Cutler

▲ PSP VINTAGEWARMER

PSP *VintageWarmer* (Mac/Win; $149) is a plug-in for VST, DirectX, and MAS host applications that models an analog multiband compressor, a single-band compressor, and a brickwall limiter. In Single-Band mode, you get a shelf filter for bass and treble frequencies. Multiband mode provides independent control of bass and treble components.

VintageWarmer adds tape-saturation characteristics to your source material. You can monitor the results with the VU and PPM meters. *VintageWarmer* requires a G3/300 MHz Power Mac, 16 MB of RAM, and OS 8.6. Windows users need a Pentium II/300 MHz, 128 MB of RAM, and Windows 95. PSP; tel. 48-60-196-3173; e-mail contact@psp-audioware.com; Web www.psp-audioware.com.

▼ ELEKTRON SPS-1 MACHINEDRUM

Four percussion synthesizers gather under the hood of Elektron's SPS-1 Machinedrum ($1,100). The Machinedrum dynamically allocates a total polyphony of 16 notes among four synthesizer engines. Sixteen tracks can be assigned for sequencing any percussion source. Each track holds a single sound element, such as kick drum or snare.

The unit's TRX synthesis engine emulates Roland's TR-series drums but provides additional real-time control over the instruments. EFM creates percussion sounds using digital FM synthesis. The E12 synth is a sample-playback component featuring sounds derived from E-mu's SP1200 and Simmons electronic drums. PI gives you physical models of acoustic drums, but you can create sounds that go beyond realistic instruments. A pair of analog inputs allows you to gate and process external audio. Sixteen dual-waveform LFOs let you synchronize the modulation of parameters.

Each track offers a dedicated effects

section. Effects include amplitude modulation, EQ, a resonant multimode filter, sampling-rate reduction, and distortion. If that's not enough, the unit's Master Effects section provides echo, delay, reverb, dynamics processing, and additional EQ.

In addition to trigger pads, the unit features eight data-entry knobs used for editing and recording synthesis and effects parameters internally or transmitting MIDI data to an external sequencer. Every pattern offers step editing of Accent, Swing, and Slide settings.

Rear-panel connections include MIDI In, Out, and Thru; six unbalanced ¼-inch analog outputs; a pair of unbalanced ¼-inch analog inputs; and a ¼-inch stereo headphone jack. Elektron; e-mail info@elektron.se; Web www.machinedrum.com.

▼ JOEMEEK MQ1

Joemeek's MQ1 (Mac/Win; $249.99) is a PCI card and breakout box that acts as a front end for a computer's sound card. The breakout box can be seated in an available drive bay.

The MQ1 uses Joemeek's compressor and EQ technology by including a 3-band Meequalizer and an optical compressor.

The box offers a breakout XLR connector and provides 48V phantom power for use with condenser mics. You also get a stereo pair of balanced ¼-inch inputs and outputs and a ¼-inch instrument-level input on the front panel. Peninsula Marketing, Inc.; tel. (310) 373-9129; e-mail themeekman@joemeek.com; Web www.joemeek.com.

▲ MARK OF THE UNICORN

MOTU's *Digital Performer* 3.1 (Mac, $795; crossgrade, $395; upgrade from *Performer,* $295; upgrade from 3.0, free) adds support for FireWire video along with a new collection of audio and MIDI features. You can use the LCD of a standard FireWire video camera or an external monitor to view your video file if the device outputs NTSC or composite video (for a review of version 3.01, see p. 120).

In version 3.1, a new window lets you globally specify that the display shows one of four time formats (measures, SMPTE, absolute time, or samples) with a single mouse-click. The update also provides an unlimited undo and redo history with a timeline and gives you random access to any edit. All edits are time stamped, and subsequent actions create a separate branch of undo events.

Digital Performer's Sequence Editor window now lets you grab and trim edges of multiple audio regions and perform destructive edits to surround audio using the Waveform Editor. You can also have imported audio files automatically conform to a sequence's tempo.

New MIDI enhancements include the ability to quantize a wider variety of MIDI events; for instance, you can quantize Pitch Bend, Program Change messages, or Modulation. The program's new Tap Tempo Entry feature lets you continuously alter tempo by tapping your controller or ASCII keyboard.

Version 3.1 adds support for Pro-

pellerhead's REX 2.0 sample-file format. You can now audition and drag and drop files created with *ReCycle* 2.0 directly into *Digital Performer* 3.1. Other enhancements include a rectified audio waveform display, transposition of Polyphonic Aftertouch events, and the ability to reassign continuous controller data in real time.

Minimum system requirements for *Digital Performer* 3.1 are any Power Macintosh with at least 128 MB of RAM and OS 8.5.1. Mark of the Unicorn, Inc. (MOTU); tel. (617) 576-2760; e-mail info@motu.com; Web www.motu.com.

▲ CAKEWALK

Version 1.3 of *Sonar* (Win, $479; upgrade, free) adds new editing and mixing tools. Editing enhancements include improvements for MIDI and audio features.

Audio Scaling enables you to create detailed edits on quiet audio passages without changing the size of the track display. A new Extract Timing interprets audio data to create tempo maps. The Scissors tool lets you split MIDI and audio tracks on the fly.

New dual-mode controls let you adjust gain and pan settings. You can now make changes in audio envelopes without disrupting audio output. Version 1.3 also provides a wider assortment of curves and crossfade types for the Fade Tool.

The program requires a Pentium II/

400 MHz computer, 64 MB of RAM, and Windows 98, ME, or 2000. Cakewalk; tel. (888) CAKEWALK or (617) 423-9004; e-mail sales@cakewalk.com; Web www.cakewalk.com.

▼ IK MULTIMEDIA
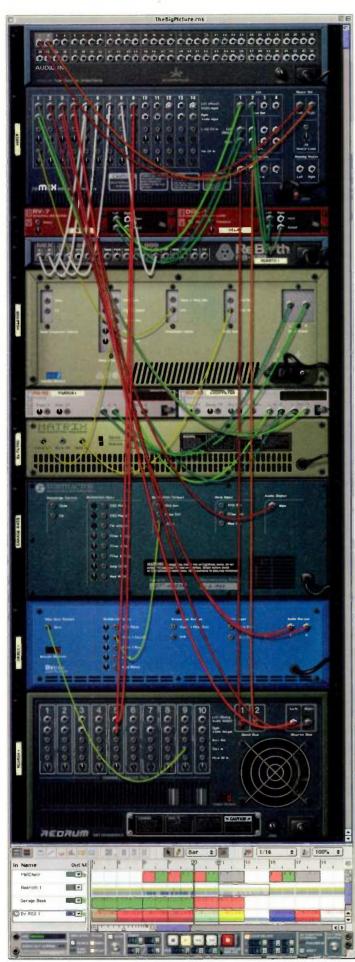
Optimized for Apple's Velocity Engine, *SampleTank* 1.1 (Mac/Win, $499; upgrade from version 1.0, free) supports dual processors for Mac and Windows hosts. *SampleTank* 1.1 for MAS easily integrates with Mark of the Unicorn's *Digital Performer.*

Among the synth's new features is a resampling and pitch-shifting engine. *SampleTank* now provides two performance modes. High Performance mode gives you 128 notes of polyphony (CPU dependent). High Quality mode reduces polyphony to 90 notes but improves the sound quality of the playback engine. IK Multimedia claims that *SampleTank* 1.1 has a markedly flat frequency response at the high end and virtually immeasurable aliasing noise. The program's Akai format converter offers assignable parameter controls for attack, release, filter-frequency cutoff, and tuning for imported Akai files.

A bug that created hanging notes in an earlier version of *SampleTank* has been repaired. IK Multimedia has also eliminated problems related to the simultaneous use of delay, autopan, and lo-fi effects. IK Multimedia; tel. (866) 243-1718 or (561) 466-9763; e-mail info@sampletank.com; Web www.sampletank.com.

▶ E-MU PROTEUS 2500

The Proteus 2500 ($1,695) is a sample-playback synthesizer module from E-mu. The 4U synthesizer offers robust real-time editing, an extensive built-in sequencer, and direct connection to USB-equipped host computers.

The Proteus 2500's ROM has a 32 MB sound set of 16-bit, 44.1 kHz samples. You can expand the unit's ROM to as much as 128 MB with E-mu's Proteus 2000 expansion ROM kits or Flash memory boards imported from E-mu Ultra samplers. Programmable knobs on the Proteus 2500's front panel let you tweak voice parameters and control Pan and Volume of as many as 32 MIDI channels in real time. For tempo-based modulation, you can sync more than 100 synthesis parameters to MIDI Clock. Sound-shaping tools include 50 filter types; 3, 12-stage envelope generators per oscillator; and 17 LFO waveforms.

The Proteus 2500 boasts improved MIDI response with a new processor that E-mu claims offers three times the power of the Proteus 2000. Audio output is also improved with 24-bit digital-to-analog converters.

A built-in 16-track MIDI sequencer offers 16 MIDI channels per track, and you can use linear or pattern-based recording. The sequencer has real-time, step, and analog-style grid recording modes, with a generous 384 ppqn recording resolution. Additionally, you can export or import Standard MIDI Files, and the sequencer section is able to hold as many as 300,000 MIDI events. The Proteus 2500 ships with a large variety of MIDI sequences that are intended for user modification.

The Proteus 2500 has six unbalanced ¼-inch analog outputs, two of which are Sub outs that double as inserts. A coaxial S/PDIF jack is used for digital output. E-mu/Ensoniq; tel. (831) 438-1921; e-mail info@emu.com; Web www.emu.com.

▼ AKAI Z4 AND Z8

Akai's Z4 and Z8 ($1,499 and $1,999, respectively) offer 24-bit, 96 kHz sampling with a 4-channel effects processor. Both have 20 GB hard drives, and you can expand sample RAM to 512 MB. The Z8 provides a removable front panel and eight real-time control knobs; the Z4 has four control knobs.

The samplers have 192 filter types, 26 of which offer resonance parameters. The filter's 6-pole design lets you set up various configurations, such as three independent 2-pole filter blocks. As many as three filters can simultaneously process a single oscillator.

Akai's Intellisample feature automatically provides keymaps for incoming audio data. A user-definable threshold triggers the sampling process; sampling ceases when the incoming signal drops below the threshold. Both samplers automatically name and assign keymaps to the sample and then repeat the process when the next piece of audio crosses the threshold.

The Z4 and Z8's built-in 4-channel, 24-bit, 96 kHz effects processor allows you to resample and process sounds. Other sample-processing features include Insert Sample, Delete Section, and BPM Match. Rescale is similar to the unit's normalizing feature, but it allows the user to define the amount of amplitude change.

The Z4 and Z8 have 248-by-60-pixel graphic LCDs and offer a built-in USB-to-host port. Akai's *ak.Sys* software lets you network and manage sampler files. Both units offer two balanced ¼-inch analog inputs and two balanced ¼-inch analog outputs. Akai Musical Instrument Corporation; tel. (800) 433-5627 or (817) 831-9203; e-mail info@akaipro.com; Web www.akaipro.com.

▶ MIDIMAN SURFACE ONE
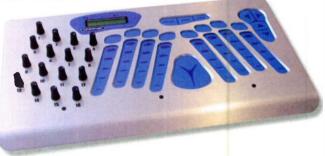
Incorporating touch-sensitive fiber-optic Smart Fabric from Tactex Controls, MIDIMan's Surface One ($799) offers an ergonomic, user-programmable desktop control surface for just about any MIDI-compatible device. The Smart Fabric material can track movement in three dimensions simultaneously—front to back, left to right, and pressure—at a high resolution. That ability lets you control multiple modulation parameters at the same time.

You can connect Surface One to a computer using the MIDI ports or the USB connector, or it can be used as a standalone control device. Surface One includes 8 fader pads, which you can subdivide into 5 triggers; 16, 360-degree rotary encoders; and 8 virtual buttons. The thumb pad is assignable as three triggers or as a single control area.

Surface One ships with presets for a variety of sequencers, software and hardware synthesizers, and lighting systems. Flash memory allows you to upgrade the unit's software. Interface connections are two MIDI In and Out ports and a USB jack. MIDIMan; tel. (626) 445-2842 or (800) 969-6434; e-mail info@midiman.net; Web www.midiman.net.

▶ CAKEWALK GUITAR TRACKS PRO

Cakewalk designed the user interface of *Guitar Tracks Pro* (Win, $139; upgrade from *Guitar Tracks*, $49; upgrade from *Guitar Studio*, $29) to resemble a portable digital studio. The software offers as many as 32 tracks and supports 24-bit, 96 kHz audio. The software does not provide MIDI sequencing.

You can use 16 simultaneous DirectX effects, with as many as four effects per track. Effects include pitch shift, parametric EQ, stereo chorus, and reverb. The program includes Alien Connections' *ReValver SE*, a guitar-amp simulator plug-in, and *Audio FX 1*, a dynamics processor. Another amenity for guitarists is *Guitar Tracks'* built-in chromatic tuner. Each of two auxiliary buses provides pre- or postfader settings.

A Master section gives you access to common global track operations such as Fast Forward, Rewind, and Record. The Track Bounce command lets you bounce any or all of the tracks to free up disk space. Multitrack editing functions include cut, copy, and paste.

Additionally, you can create volume and pan envelopes.

The software ships with a large number WAV-format loops compatible with Sonic Foundry's *Acid*, and you can import them into the program with automatic tempo matching. You also receive *Fruityloops Express* for creating your own drum loops.

In order to run *Guitar Tracks Pro*, you will need to have at least a Pentium II/300 MHz computer with 64 MB of RAM and Windows 98, 2000, ME, or XP. Cakewalk; tel. (888) CAKEWALK or (617) 423-9004; e-mail sales@cakewalk.com; Web www.cakewalk.com.

▶ UNIVERSAL AUDIO NIGEL

Universal Audio has released *Nigel*, a guitar-amp modeling plug-in for its Powered Plug-In digital signal processing system. The software emulates a range of revered amplifiers, delivering everything from clean, warm tube sounds to replicas of saturated British amps. *Nigel* also allows you to create nontraditional amplifier tones.

The plug-in lets you morph between any two amplifier presets and features a bevy of stompbox emulations. Effects include phase shifters modeled after the Mutron Bi-Phase, the Electro-Harmonix Small Stone, and MXR phasers; wah effects

patterned after the Mutron III; a gate and a compressor; and fade-in, tremolo, and delay effects.

Nigel is free to all Powered Plug-In owners, and the software will be included in an upcoming Powered Plug-Ins 2.0 update. Perhaps the update will include knobs that go to 11. Universal Audio; tel. (831) 466-3737; e-mail info@uaudio.com; Web www.uaudio.com.

▶ BLACET MINI-WAVE
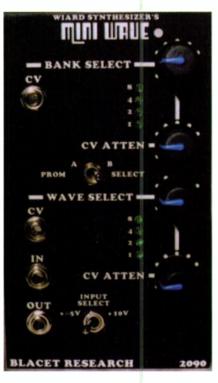
The Mini-Wave (assembled, $239; kit, $180) is a hybrid analog and digital synthesizer module from Blacet. Grant Richter of the Wiard Synthesizer Company designed the module, and Blacet retooled it to accommodate the Paia FracRak mounting format. The Mini-Wave stores a 512 KB digital wavetable with 256, 8-bit sampled waveforms arranged in 16 banks. The module can be used as a sound source or function generator.

You can select any bank of waves with the Bank Select knob and scan or select waves manually with the Wave Select knob. Other possibilities include scanning the wavetable and banks using control voltages (CVs) or a joystick. A row of LED indicators illustrates the bank and waveform selection in binary. The Mini-Wave includes an Input Select switch for changing the CV to 0V through 10V, or ±5V.

All inputs and outputs are on ⅛-inch minijacks. Besides audio I/O, there are CV inputs for Wave Select and Bank Select. Power supply and rack housing are sold separately. Blacet Research; tel. (707) 869-9164; e-mail blacet@blacet.com; Web www.blacet.com. ✪

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SOUND CHASER

VINTAGE PAGE

By Julian Colbeck

Akai S950

Produced: 1988–93
Made in: Japan
Designed by: Dave Cockerell, H. Takemura
Number produced: 15,000
System: 12-bit linear sampling
Price new: $2,500

Today's prices:	*Like new*	*$500*
	Like, it's okay for its age	*$425*
	Like hell	*$300*

When the S950 was introduced in 1988, all eyes were focused on Akai's more newsworthy sampler, the S1000. The S950, essentially an upgraded version of the popular S900, hardly rated mention in the popular press.

It wasn't long, however, before many sampler-savvy musicians realized that the S950 was quite well suited for a variety of musical applications. Although the S950 offers the same meager 12-bit processing and 8-note polyphony as the S900, it boasts a superior 48 kHz sampling rate and a maximum 2.25 MB of RAM, with 750 KB standard. In addition, items that were options on the S900, such as crossfade looping and pretrigger recording, are standard on the S950. The S950 provides 99 sample locations as opposed to only 32 on the S900.

Physically, an S950 is almost indistinguishable from an S900. It has the same green LCD, the same eight Page buttons, the same cursor, and the same five knobs. Although the disk drive looks identical to the S900's, the S950 accepts both double-density (DD) and high-density (HD) floppy disks. You can load disks while playing, too—a feat the S900 is not inclined to perform. Also unlike its predecessor, the S950 provides double-speed MIDI communication for transferring data to a sample-editing program on an external computer.

The S950 operates much like an S900 does: the sample record pages take you through setup procedures one step at a time. Although the lofty 48 kHz rate offers extra sparkle, it also reduces your maximum sample time at full bandwidth.

Using the S950's pretrigger recording, you can record razor-sharp attacks. If you really want to get serious, you can process individual hits from the S950's eight outputs.

The S950 was the first Akai sampler to implement time stretching, enabling you to alter sample length without altering pitch. Considering the S950's continued popularity as a drum-loop sampler, that capability proved crucial to its success. If you stray too far from the original sample length, recordings begin to sound a little metallic and artificial, which might be the result you're aiming for.

The S950 has stood the test of time well; many people have continued to use theirs through the 1990s and beyond. For Steve Hackett's album *There Are Many Sides to the Night* (Caroline Records, 1994), I spliced a scratchy old solo violin sample from the S950 onto the attack of a lush strings patch from the Korg Wavestation. Perhaps more resourcefully, Norman Cook (aka Fatboy Slim) uses a transposed S950 Tone Program mixed in with an Oberheim Matrix-1000 for many of his subbass sounds. S950 drum loops—grainy and grungy in all their 12-bit glory—are particularly hip in hip-hop.

A vast range of Akai and third-party sounds are available for the S950. Impressively, the S950 is downwardly compatible with S900 disks and upwardly compatible with S1000 disks. Although the S950 converts the S1000's

An instrument beloved by users but initially ignored by the media, the S950 has become a classic vintage sampler, with all the physical directness and audio graininess that such a distinction implies.

PRICE GUIDE

The quoted prices reflect typical street prices you must expect to pay in U.S. dollars. The buy-in on vintage instruments, as with vintage cars, is just the beginning, though. Most of the original manufacturers are long gone, so maintenance and repairs are expensive.

16-bit data to 12 bits and stereo to mono, most samples transfer extremely well.

The S950 has a few known problems. If the instrument is handled roughly, the EXM006 expansion chips are somewhat prone to falling out; luckily, it's easy to remove the bottom plate and press the errant chips back into place. The disk drives have been known to wig out eventually, and replacing them costs as much as $200.

Here's a tip worth mentioning: if an S950 disk appears damaged and you fear that your sample data is irretrievably lost, try loading the samples or programs individually instead of powering up with the disk in place. That trick will get you out of jail half of the time.

The S950 was never a fancy machine with advanced facilities such as onboard effects, stereo sampling, resonant multimode filters, and the like, but it's a workhorse. In the S950's prime, SCSI was available as an option. Numerous software programs can transfer samples to and from your Macintosh or PC for editing.

The S950 offers a particular coloration and feel, and it's quite capable of filling a niche in the 21st-century studio. The Web is dotted with S950 info, and you can even find some resources at Akai's site (www.akaipro.com), including a searchable PDF version of the S950 manual. In addition, Masterbits (www.masterbits.com) offers a 750 KB Memory Expansion Card from the German company MU-TEC.

Julian Colbeck has toured everywhere from Tokyo to São Paulo with artists as varied as Yes, Steve Hackett, John Miles, and Charlie.

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

By Matt Gallagher

Pleasant Exchange

A Twin Cities band builds two home studios and enjoys good equipment karma.

Pleasant Stitch's music defies precise categorization. Touching on ambient, dark-wave pop, and gothic trip-hop styles, it is influenced by the Cocteau Twins, Love and Rockets, Kraftwerk, and Art of Noise, among others. Pleasant Stitch is Bob DeMaa, Carty Fox, Sarah Jane Hill, and Will Pierce. Their self-titled debut album is an ethereal work laden with synths, piano, textured guitars, and driving electronic percussion, topped with Hill's expressive vocals.

The band recorded *Pleasant Stitch* in three studios during a two-year period. "I could never imagine paying somebody to record my stuff for me because no one's going to care about it as much as I do," says DeMaa, "so I bought a Pro Tools rig and went to a tech school."

Recording sessions began in a St. Paul house shared by DeMaa, Fox, and Pierce. "It was a duplex, and Carty lived next door," says Pierce. "Bob was in the basement, and I was upstairs." DeMaa set up his Pro Tools III system on a Power Mac 7600/120 MHz in the basement, along with a Mark of the Unicorn (MOTU) MIDI Time Piece AV; API, A.R.T., and Mackie microphone preamps; and an Audio-Technica AT4050 condenser mic for the vocals. The house was strewn with synthesizers. "The equipment list is extensive because we buy and sell gear—often," says DeMaa, noting that the group also used several software synths and sample CDs.

Band members experimented with tracks individually and exchanged MIDI and audio files separately. Fox and Pierce each had a Mac running Steinberg's *Cubase VST*. They subsequently replaced *Cubase* with Emagic's *Logic Audio* at DeMaa's behest. "I like to mix in *Pro Tools*, but I prefer *Logic* for writing," DeMaa says. The entire project moved to a house in Minneapolis when Fox got married.

One characteristic of the album is its creatively mutated string patches. For example, on the track "Beggar," Fox rearranged the original string and piano parts "in almost a 12-tone, progressive manner over 10 minutes. It also has odd samples halfway through it. I think I grabbed those arbitrarily and then hacked them and whacked them in [BIAS's] *Peak*, which is one of my favorite things to do. I did that to [Hill's] vocals on 'Dandelion Heart.'"

Hill's singing set the stage for "Beggar" and other tracks. On "Beggar," Fox says, "her vocals were so gorgeous that we just took all the music out and restarted from scratch, mainly around the vocals."

DeMaa works full-time as an engineer at a Minneapolis commercial studio, Asche and Spencer, which served as the band's third studio. "I modeled all the Pro Tools/24 studios here after my own," DeMaa says of Asche and Spencer. "For the final stages of the record, I basically had four exact duplicates of my own studio, and we were all able to work on things at the same time, in different rooms. We finished up the mixing and mastering in here."
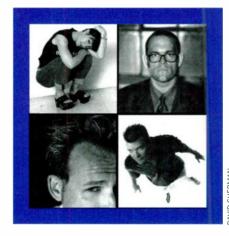
"We started a habit that we've continued to this day," Fox says, "which is we each work at home on separate PowerBook systems. We swap files. We have a common FTP spot on the Internet where we keep the main sessions, and we download them, work on them, and reupload them."

"At the end of the record, we figured out how to work," says Hill. "Sometimes with the first album, it seemed like we were all swimming in different oceans, trying to find each other." ⏺

For more info, contact Pleasant Stitch; e-mail mail@pleasantstitch .net; Web www.pleasantstitch.com.

DAVID SHERMAN

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BY ZACK PRICE

Anyone who has played Tascam's *GigaSampler* or *GigaStudio* (originally developed by NemeSys Music Technology) readily understands why it is rapidly becoming the de facto standard for PC sampler users. Because *GigaSampler* plays samples from a hard disk rather than from RAM, it doesn't suffer from the same memory-storage limitations that plague RAM-based samplers. That means you can use full-length samples without loops to better reproduce an instrument's sound. Granted, some situations call for looping samples in *GigaSampler*, but the looping doesn't compromise the sound quality or realism, because the loops can be so long they're virtually undetectable. Moreover, *GigaSampler* isn't bound by the same constraints as other samplers regarding the number of samples that constitute an individual multisample Instrument (see the article "Soft Sampling" in the October 2001 issue).

Want to make great *GigaSampler* libraries? Begin with a solid pre-production plan.

Combine the freedom to create a vast number of full-length, high-quality samples with *GigaSampler*'s extensive sample organization and MIDI control parameters, and you have the capability to create and play sampled instruments that reproduce with astounding accuracy the nuances of the actual instruments. Although this massive instrument-design potential may be liberating, it can also be intimidating to the neophyte *GigaSampler* user. The program's supplied tutorial shows how to create simple Instrument libraries but doesn't demonstrate how to create more complex libraries like those that are available commercially in Giga-Sampler format. What's more, the tutorial doesn't provide instruction in the all-important pre-production phase of development.

Library Science

Pre-production is the key to creating an efficient and well-organized *GigaSampler* library, whether it's for private use or for commercial release. Beginners, however, often bypass pre-production and jump right into the sample-recording stage of development without clearly mapping out their goals or anticipating roadblocks. Instead of wasting time making mistakes, you can learn the secrets of effective pre-production planning by examining the approaches taken by professional sample-library developers. You may even find that mastering *GigaSampler*'s potential is easier than you thought, as long as you're willing to properly lay the groundwork with a solid pre-production plan.

CLIMBING THE DECISION TREE

The first step in the pre-production design process is to gain a clear and detailed picture of what you want to sample: a specific instrument, a collection of instruments, loops, or sound effects. Your decision at the initial stage will determine the approach you take for the whole project, so it is very important to plan carefully (see Fig. 1).
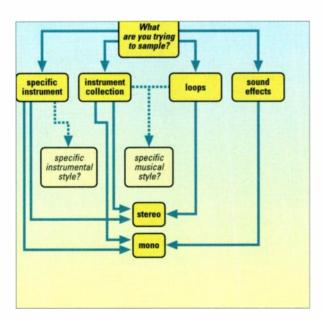
If you want to create a guitar library, for example, you could design one that concentrates on reproducing individual notes from a specific electric, steel-string folk, nylon-string classical, or 12-string guitar. Likewise, you could design a library that includes an assortment of different types of guitars or loops of guitar riffs.

What you choose to include in your library determines its eventual production requirements. For example, specific instrument libraries are usually highly detailed, capturing as many nuances of the instrument as possible. That means you will need a lot of samples and probably a complex design structure. Instrument collections, on the other hand, may not include all the playing nuances of the single-instrument libraries but may still require a large number and variety of samples with a different kind of organization.

Jim Corrigan's *Nashville High-Strung Guitars,* for example, focuses on recreating an acoustic-guitar sound employed in country-music arrangements. Larry Seyer's *Upright Acoustic Bass* reproduces the plucked sound of an upright bass for jazz, bluegrass, swing, and other nonclassical styles.

You can also focus on a particular playing style but apply it to a collection of instruments in the style of Christian and Lane's *Orchestral Percussion,* Sonic Implants' *Afro-Cuban Percussion,* and Jim Corrigan's *Celtic Instruments.* Loop libraries are even more focused on a particular musical style, as evidenced by CDs such as Numerical Sound's *Sly Dunbar Reggae Drum Splash* and Bigga Giggas' *Harmonica Essentials: Blues Licks.*

If you want to sample a specific instrument, you may find that narrowing your focus and reducing the variety of sounds can yield a higher-quality result. For instance, Peter Ewers's *Symphonic Organ Samples* library uses only five of the most common organ-stop combinations on a historical French-cathedral instrument. By concentrating on a few no-compromise patches, he was able to deliver an excellent rendition of the instrument complete with dedicated reverb tails from the cathedral itself. If he had tried to incorporate all the possible organ-stop combinations, producing or selling the CD-ROM wouldn't have been feasible. Likewise, Seyer excluded bowed sounds from his *Upright Acoustic Bass* library because that playing technique did not fit with the intended musical 40

MANO A MONO

Before you move on to the next stage of pre-production, decide if you want to record your samples in mono or stereo. Your decision should be based on a number of aesthetic and technical factors. Although it seems that everyone samples in stereo, sometimes it's preferable not to. For example, Bigga Giggas' *Rhodes for GigaSampler* and *Hohner D-6 Clavinet* use mono samples because the original instruments had mono outputs. Loop libraries are almost always stereo because loops are often the bedrock of

FIG. 2: This is a detail of the Dimension Assignment area in *GigaEdit* 2.0. The samples in this region are in stereo (one bit), arranged in four Velocity levels (two bits), with four stereo sample layers (two bits). All five Dimension bits are therefore assigned.

FIG. 1: Before you begin creating a sample library, it's important to ask the right questions, plan ahead, and devise a workable strategy.

an arrangement. Sound effects are usually recorded in mono, so the end user has more control over the placement and movement of the sound in a stereo or surround-sound environment.

Corrigan also suggests examining an instrument's sound-radiation pattern to determine whether its samples should be mono or stereo. For instance, a trumpet might be sampled in mono because its sound-radiation pattern emanates from the bell and is therefore more localized. On the other hand, an instrument with a more complex radiation pattern, such as a guitar or piano, will probably sound better if sampled in stereo. If you're trying to capture the sound of the trumpet and the room in which it is played, however, stereo sampling is a better bet.

Corrigan broke his own rule when he created his *Nashville High-Strung Guitars* library. He used separate mono recordings of the low-voiced and high-voiced guitars panned hard left and right, because that's how the sound is recorded with live players in Nashville studios. As you can see, the musical context alone might determine the final approach.

A couple of technical considerations may also influence your decision to create mono or stereo samples. If you plan to create a GigaSampler Instrument by layering samples, keep in mind that *GigaSampler* lets you layer two stereo or four mono samples per note. (*Giga-Studio* allows you to layer four stereo or eight mono samples per note.) If you know ahead of time how many layers you'll need, you can decide if sampling in stereo is necessary or even possible.

In addition, *GigaSampler* uses one of its Dimensions (sample control sources) just for stereo samples. The program lets you set the resolution for each of its five possible Dimensions (seven in *Giga-Studio*) up to a total of five bits (or 32 splits). For instance, mono samples don't use any bits, but stereo samples use one Dimension bit. That leaves four bits (16 splits) to be divided among the remaining Dimensions. (The number of splits drops exponentially.)

In addition to the bit used for stereo, two bits (four splits) are used for the four Velocity levels, and two bits are used for the four sample layers (see **Fig. 2**). That effectively uses up all the possible Dimensions. If you suspect that you might need to use many Dimensions for other purposes, be sure to use mono samples.

THE NEXT STEP BEYOND

The next pre-production step involves analyzing how a performer actually plays the instrument in the style that you're trying to capture. That is important because you must generate a performance model that you can later apply to various MIDI controllers. Moreover, the model will influence how you design the *GigaSampler* "translation" of the instrument, and it will give you an indication of the types and numbers of samples that you'll need.

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Strummed instruments, such as guitars, ukuleles, and banjos, may need a variety of sampled chords that are switched with trigger keys. Plucked instruments may require string-bending or damping effects; acoustic and electric pianos rely heavily on Velocity control and a sustain pedal.

A pipe organ, on the other hand, is more like a wind instrument; its sound doesn't decay into silence like a piano's. For greater realism, you have to loop a sampled organ's sound and use separate release samples to imitate the action of stopping the bellows from blowing air into the corresponding pipe.

MAKE MINE MIDI

Once you have determined your instrument's behavioral model and have selected which of its articulations you want to capture, you must choose the appropriate MIDI controllers for your instrument. Naturally, you will have to set up your library to work with keyboard controllers, because they're the most widely used type. However, other controllers may work as well or even better.

For example, Seyer's *Upright Acoustic Bass* has an Instrument that lets keyboard players "switch strings" while playing by using trigger keys. The CD-ROM also includes four Instruments, one for each string, that you can load into separate MIDI channels to play with a MIDI guitar or bass controller for a more natural playing option. In a like

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MAKING NEW CONVERTS

You've just bought Tascam's *GigaSampler* or *GigaStudio* but haven't had the chance or the cash to invest in a CD-ROM library, and you're not ready to dive into creating your own sample library from scratch. You also have a boatload of samples in other formats that you want to use in *GigaSampler*. What do you do? Fortunately, *GigaSampler* and *GigaStudio* include a program called *S-Converter* that lets you convert Akai S1000 and S3000 files on CD-ROMs to Giga-Sampler format. *GigaStudio* also includes *A-Converter,* which lets you rip tracks from audio CDs so you can create loop and sound-effects libraries in Giga format more easily. In addition, *Instrument Editor* 2.0 in *GigaStudio* lets you load Sound-Fonts 2 (SF2) and Microsoft Down-loadable Sample (DLS) files for easy conversion to Giga-format files.

Several third-party programs can convert between different sampler formats including GigaSampler. Chicken Systems' *Translator* 2.1 ($149.95; www.chickensys.com) and Amazing Sounds *CDxtract* 3.5.1 ($79.00; www.cdxtract.com; see Fig. A) convert a variety of sample file types, but they read only CD-ROMs or files on removable and fixed hard drives. Neither reads sample files from floppies. It is, however, possible to find utility programs that can put floppy-disk images onto a hard drive. *Translator* even provides those free utilities as part of its full package. Both *CDxtract* and *Translator* come in GigaEdition packages ($79.95 and $59.95, respectively) that translate a variety of sample formats into just GigaSampler format.

Another file-conversion program with a long history is FMJ Software's *Awave Studio* 7.3 ($89.95; www.fmjsoft.com). Not only does it convert a variety of sample formats but it also reads and converts numerous tracker program files, disk images of various Ensoniq synthesizers, and a variety of obscure audio-file formats. Unlike *CDxtract* and *Translator, Awave Studio* only reads CD-ROMs that are in ISO9660 format. It can, however, read floppies if they're in MS-DOS format.

All of the conversion programs do a fine job of converting sample libraries, and all use the Windows Explorer–style user interfaces. However, no one program is capable of translating all file formats, so check each company's Web site for updated file-conversion information.

FIG. A: Like other sample-conversion programs, Amazing Sounds *CDxtract* 3.5.1 uses a Windows Explorer–style interface to view and convert samples from CD-ROMs and hard disks.
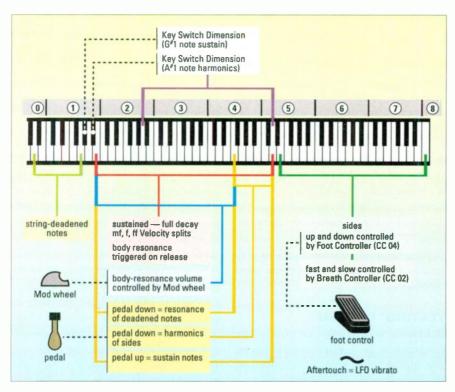
FIG. 3: This diagram of the Optimal Upright Bass Instrument from Larry Seyer's *Upright Acoustic Bass* shows how the samples are mapped to MIDI notes and how Dimension assignments are made for key switching, MIDI Sustain, Mod Wheel, and other functions.

manner, percussion samples often produce more natural-sounding percussion tracks when played with a percussion controller.

LIVE OR MEMOREX?

Once you have established which controllers you will use, you must determine whether it is possible to play your Instrument in a way that generates all of its nuances in real time. For some instruments, such as pianos and percussion, the answer is a definite yes. For other instruments, you may have to assign different articulations to separate Instruments and load them into *Giga-Sampler* at different Levels. (Levels let you layer patches.) For example, the *gadulka* (Bulgarian fiddle) sound included in Quantum Leap's *Rare Instruments* CD contains separate Gadulka and Gadulka Resonance Instruments. To reproduce the authentic sound of the instrument, you load Gadulka into Level 1 and Gadulka Resonance into Level 2 on the same MIDI channel to hear both the bowed and sympathetic strings.

Sometimes it isn't possible to incor-porate all the desired articulations into a single *GigaSampler* Instrument. In that case, you may have to create separate Instruments for the extra articulations. For instance, Thomas Skarbye's *Scarbee J-Slap Bass* crams a huge assortment of articulations into its Studio Bass multi-sample. Even so, separate Instruments were created for slides and fingered harmonics that have to be accessed through patch changes. You might be able to play and generate patch changes quickly enough to play all the articulations in real time, but in most cases, you'll have to take a couple of recording passes in your sequencer and perhaps do a little editing as well.

Another alternative, which *Scarbee J-Slap Bass* employs in its Jam Bass patch, is to create an Instrument that has fewer articulations but that still offers a realistic sound. Seyer's *Upright Acoustic Bass* has an Instrument named Optimal Upright Bass that takes the best of the capabilities of the Switchable String Position Bass to make it easy to play in real time (see **Fig. 3**). It's easier for the performer to use this patch because it

lacks the extra element of key switching to move string positions.

Determining whether it is possible to play your instrument in real time implies that you have started the process of assigning articulations to Dimensions. It isn't as complicated as you might think. You should have already determined whether the samples are to be mono or stereo, so that's one Dimension you have assigned. You may not know exactly how many Velocity levels (second Dimension) you will eventually use, but a maximum of four usually is sufficient for most projects. Depending on the instrument, you may not need Velocity splits at all; that would leave you with even more room for other Dimensions. Other common Dimensions are MIDI Sustain and Mod Wheel.

Keep in mind that assigning Dimensions is preliminary at this stage; you might change some Dimension assignments later. However, remember two things. First, you should document what you do and update that documentation as you change things. Second, you should try to assign Dimensions in a way that makes sense. For example, don't have the Mod Wheel control the sustain samples when the Sustain Pedal is a better choice. *GigaSampler* guru David Govett recommends that you assign Dimensions to the available features of most MIDI controllers rather than assigning them to controls that you may have but which are not commonly used by other people.

TESTING, TESTING

Believe it or not, you are now at the final stage of pre-production, where you are about to put your preliminary library design to the test. Before you begin sampling, make sure your instrument is in tip-top shape and that the instrument and related equipment don't have mechanical or electrical problems. You may also find that it is necessary to take extra steps to improve upon the instrument's natural sound. As an extreme example, Corrigan developed a program to generate open tunings using different string gauges to eliminate fingered string buzzing in his *Nashville High-Strung Guitars* library. Nick Phoenix, producer of Quantum Leap's *Rare Instruments* library, used a Roland VP-9000 processor to stretch the sound of a *ragdung* (10-foot Tibetan trumpet) in real time as it was sampled.

Now record some test samples of the instrument and some of the articulations you wish to capture. Listen to the samples to check for flaws inherent to the instrument that may not have been obvious when listening with the naked ear. Jim Van Buskirk (founder of NemeSys Music Technology) recommends that you create and play a mini-Instrument

FIG. 4: This is a recording template created by Jim Corrigan in *Cubase VST/24* 5.0 for recording and cataloging the major seventh chords used in *Nashville High-Strung Guitars*. The chord names are assigned to Tracks, and each articulation for that chord is a Part in a particular Track.

with a three-octave range so that you can test the samples and the preliminarily assigned articulations together in context. Use this alpha test to determine what's right about the Instrument design and to isolate any problems.

Once you've run some test samples and you're ready to start sampling in earnest, create a recording template similar to the one shown in **Fig. 4**. That example comes from the session work for *Nashville High-Strung Guitars*. Corrigan used 12 tracks to indicate the 12 major-seventh chords he needed to record. On each track, he indicated the different samples he would need for that particular chord. Reading from left to right, he inserted a part for Left Guitar Upstroke, Left Guitar Downstroke, Left Guitar Damped, Right Guitar Upstroke, Right Guitar Downstroke, and Right Guitar Damped. In a similar manner, you should give the tracks and parts in your template names that reflect, in an organized manner, the samples that you want to record. It's important to have a template that moves the sampling sessions along in an efficient and logical manner and to have consistent sample names that are easily identifiable.

JUST DO IT

You are now ready to move into the production phase. Don't forget to name samples logically and to use MIDI note names or numbers at the end of the sample file name. As stated in the *Giga-Sampler* tutorial, the Instrument Wizard uses that information to speed up the process of mapping samples in the Instrument-creation process. In addition, remember to create and appropriately name folders so you can more easily organize and access samples by type. Afterward, you should be able to create your Instrument using the Instrument Wizard to lay out sample maps and the Dimensions to control how those samples interact when you play the Instrument. Good luck; I hope to hear a great *GigaSampler* library from you soon.

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

2002 EDITORS CHOICE

Three cheers for our 33 favorite new products of the past year.
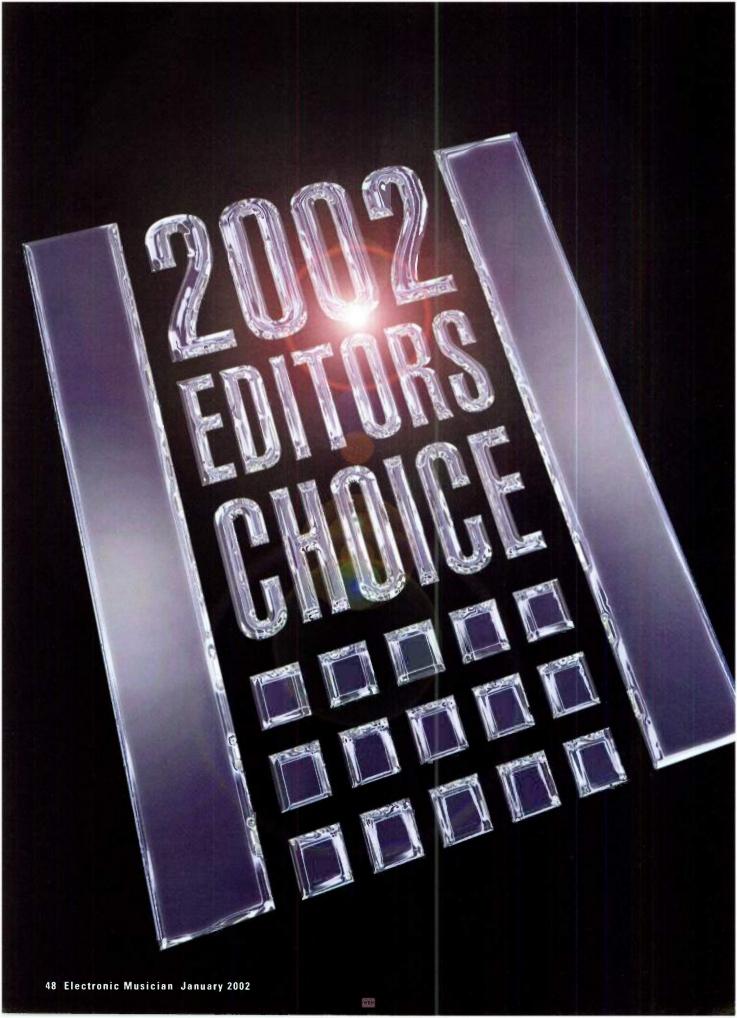
In December 1992, EM's editorial staff—Bob O'Donnell, Michael Molenda, Steve O, and Scott Wilkinson—unveiled the first Editors' Choice Awards. We gave our highest accolade to 14 hardware and 6 software products, including 2 whose descendants triumphed again this year: Digidesign's Pro Tools (now *Pro Tools TDM* 5.1) and Twelve Tone Systems' *Cakewalk Pro* for Windows (which evolved into Cakewalk's *Sonar XL* 1.02). Other now-legendary 1992 winners included the original Alesis ADAT, Steinberg's *Cubase Audio* 1.01, and the Kurzweil K2000.

The industry obviously has evolved considerably in the ensuing ten years and so have the awards. Every year we have added to or changed the prize categories slightly to reflect the types of products we felt were significant for that year. This year we've selected a record 33 products in 31 categories, of which an unprecedented 7 categories are for various types of synths and samplers (5 software and 2 hardware). We also give honors in three software and four hardware signal-processor categories. Add the Sound Design Software award to the list, and you'll find that a total of 15 categories—nearly half of the awards—are for sound-creation and sound-processing products. Of those, nine are for software.

Although much has changed in the past decade, it's reassuring to note that some things remain the same. As always, the latest batch of award-winning

By the EM Staff

products has been field-tested during the past year by our editors and a select group of top authors. We also solicited opinions from the editors of sister publications *Mix, Onstage,* and *Remix*. The final selections were made by our technical editors: Steve O, Marty Cutler, Brian Knave, Dennis Miller, Gino Robair, and David Rubin. We went with some of the latest stuff this year, so in several cases, we have not yet published reviews of the winning products, but we will endeavor to do so as soon as possible (see the sidebar "The Award Winners in Review" on p. 88).

Remember, these are our picks among the new products we tested during the past 12 months; we cannot consider the many fine products we never got to check out, and we do not consider older products. We bestow an award on a software upgrade only if we think it offers major and significant improvements upon the previous version.

To be eligible, the product must have shipped between October 1, 2000, and October 1, 2001. To be fair, we allow some slack for products that shipped immediately before October 2000, so close to the deadline that it was impossible for us to test them in time for last year's awards. We do not allow such slack if we believe a manufacturer could have supplied a review unit in time for last year but intentionally delayed sending it—that is, if we have good reason to think the company ducked us.

And now we are proud to present the winners of the tenth annual EM Editors' Choice Awards. Drumroll, please!

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

There 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."

Ancillary Hardware

TERRASONDE
AUDIO TOOLBOX PLUS ($1,699)

TerraSonde's Audio Toolbox Plus is one of those items that, once you use it, you can't believe you lived without it. This little wonder packs a wealth of diagnostic functions into a handheld device and

can be used to analyze a studio's acoustics as well as to test your gear. Although its price may appear steep at first, it is well below what you would pay for each component separately.

If you want to tune your room, you can use the Audio Toolbox Plus's real-time analyzer. The device can also measure sound-pressure level, reverb decay time (referenced to the RT60 standard), and the background noise in a room. TerraSonde offers Mac- and Windows-compatible software for extracting and printing the graphs created with the unit.

The Audio Toolbox Plus will also help you get the most out of your gear. Use the polarity tester to make sure your speaker components are in phase and the signal-to-noise-ratio function to measure the quality of your mixer and effects processors. The unit also provides a signal generator, an impedance meter, a frequency counter and level meter, and a distortion meter.

For the desktop musician, the Audio Toolbox Plus can read and generate the common SMPTE time-code rates and MIDI Time Code (MTC). The unit allows you to monitor MIDI activity, display MIDI information in hexadecimal or text format, test MIDI cables, or transmit MIDI messages. For more traditional studio chores, it includes a digital oscilloscope and can test cables, phantom power, and batteries. You also get a small built-in speaker, an internal rechargeable battery, and a sturdy case.

In short, TerraSonde's device easily can handle most personal-studio testing situations. If you're serious about your studio and you want to evaluate it at a pro level, the Audio Toolbox Plus is a worthwhile investment.

Ancillary Software

CHICKEN SYSTEMS
TRANSLATOR 2.1 (WIN; $149.95)

Desktop musicians who rely on samplers for music production are used to a certain amount of frustration. Perhaps you've just discovered that the perfect sample CD for your next project is not available in your sampler's format or that your extensive Kurzweil library is useless with your new E-mu sampler. As an increasing number of hardware and software samplers enter the marketplace, sample obsolescence has spread like the plague.

What electronic musicians need now more than ever is a Rosetta stone for samplers; fortunately, Chicken Systems heard their pleas and introduced *Translator*, the ultimate file-format-conversion program for Windows. Version 2.1 boasts a range of new features, but its greatest strength lies in the astounding number of sample formats (more than four dozen) that it can translate to and from, including most hardware models from Akai, E-mu, Ensoniq, Roland, Kurzweil, Korg, and Yamaha as well as software samplers from Tascam (NemeSys), Steinberg, Digidesign, Emagic, and many others.

Translator doesn't just convert raw samples from one device to another; it also converts (as much as possible) keymaps, modulation routings, Velocity switches, envelopes, tuning, loops, filter settings, and various additional parameters. What's more, you can translate a single sample, program, instrument, preset, bank, volume, or whole directory—even an entire disc.

Translator uses a Windows Explorer–style directory tree as its interface for quick and easy browsing, and you can audition any file directly from your hard disk. The program lets you create a virtual drive on your computer and format it for any supported samplers. In fact, you can batch-convert multiple sampler formats into a single format. If you select all of the contents of a sampling CD and drag the files to your hard drive, you can watch in amazement as *Translator* chugs away, converting your library in record time.

Translator has many more features for managing

sample libraries, and the program is constantly being upgraded and expanded. If you work regularly with samples, this is one piece of software you'll want to keep handy. It's the strongest weapon yet in the fight against sample obsolescence, and for that it deserves recognition.

Audio-Editing Software

DIGIDESIGN

PRO TOOLS TDM 5.1 (MAC/WIN; WITH PRO TOOLS 24/MIX, $7,995)

Pro Tools has been an industry standard for years. With the introduction of the appropriately named version 5.1 software (upgrade $295), Digidesign adds surround-sound capabilities as well as many other enhancements to the software part of its flagship system. (Note that *Pro Tools TDM* software requires Pro Tools 24/Mix hardware.)

To begin with, *Pro Tools TDM* 5.1 supports multichannel tracks; you can now record as many as eight channels of audio in a single track. The channels in a track are edited as a unit, and the corresponding regions are conveniently stored hierarchically under a master region name. In addition, you can use multichannel plug-ins with multichannel tracks. For example, Digidesign has introduced the *SurroundScope* plug-in, which shows you where a sound is placed in the surround environment, with level meters and a phase meter thrown in for good measure. Multiple instances of mono plug-ins can be controlled individually or linked together, and *Pro Tools TDM* 5.1 lets you open more than one plug-in window at a time.

Pro Tools' MIDI implementation has been beefed up considerably. The Event List is a welcome addition, as is the ability to record multiple MIDI tracks simultaneously and play a MIDI controller through the program without being in Record mode.

High on the list for electronica fans is Beat Detective. By scanning an audio file for transients, Beat Detective can extract groove information and make other files conform to it. It can also quantize

and correct timing irregularities in a file, smooth over edits, and create a tempo map.

Pro Tools TDM 5.1 is more stable and robust than earlier versions. If you already use *Pro Tools*, the upgrade is a no-brainer. If you're considering the purchase of a TDM system, the new features, as well as the program's prominence in the pro-audio world, should provide a convincing enticement to jump in.
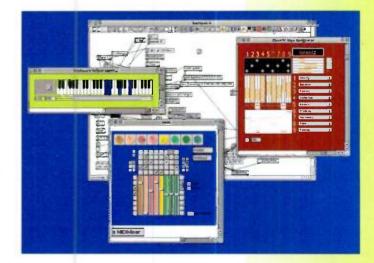
Composition Software

CYCLING '74

MAX 4.0 (MAC; $295)

Rarely does one product totally dominate its class. If you're interested in a graphical music-programming language for the Mac, Cycling '74's *Max* 4.0 will top your list. Use *Max* to create elaborate processes that transform your MIDI data in unusual ways. Send *Max* a series of chords, and it will generate endless variations on your harmonies. Play five notes in succession, and it can record the notes and loop them bidirectionally, randomly changing octaves on each pass. What better credentials to have than Kay Algorithmic Real-time Music Architecture (KARMA), the algorithmic MIDI processing section of the Korg Karma keyboard, which materialized eight years ago as a *Max* patch? (Its current implementation is in C/C++.)

Max can control almost anything with MIDI: use it to run a video deck from a MIDI controller, to play tracks from an audio CD randomly, or to change the speed of a QuickTime movie. If interactivity is your game—whether for a kiosk, in the concert hall, or on the Web—you'll find *Max* has everything you need.

Max has always been a vast resource for tweakers, but version 4.0, the first major release in nearly five years, really takes it over the top. First, you'll notice the interface's modernized look; in fact, many of the new graphical elements were created within *Max*. It's easier to manipulate objects than in the past—the ability to create a background layer, for example, keeps your desktop uncluttered as you build your patches—and the new color-coded patch cords help

keep the structure of your patches clear. Version 4.0 also has new objects for data processing. Clearly, this update has more than meets the eye.

To sweeten the pot, you can get *Max* bundled with *MSP* 2.0 (also an Editors' Choice award winner; see p. 80) for $495. Here's some good news for PC users: *Max* and *MSP* are under active development for Windows. Although no firm release date is set, working versions have appeared at major trade shows, so keep your eyes on the Cycling '74 Web site for news.

If you wonder what a full-on music-programming environment can do for you, *Max* is your guy!

DAW Control Surface

RADIKAL TECHNOLOGIES
SAC-2K ($1,849)

As music software expands beyond the tasks that were once the exclusive domain of hardware devices, the need for hands-on control is greater than ever. Using a mouse to individually manipulate onscreen sliders and knobs is far inferior to actually pushing real sliders and twisting real knobs. Fortunately, control surfaces are catching up with the need to physically manipulate the details of music production. Although many new control surfaces have been introduced in the past year, deciding on our favorite was easy.

The SAC-2K combines extensive functionality with realistic bang for the buck (for a review of the SAC-2K, see p. 146). With 9 motorized faders; 67 buttons; an assignable Shuttle wheel; 3, 80-character text displays; a separate locator display; and 12 rotary-encoder knobs with 31-segment LED rings, the SAC-2K provides a versatile tactile interface for working with virtual instruments and digital audio sequencers alike. The moving faders can be adjusted to respond to a light touch or a heavy hand. The touch-sensitive rotary encoders respond to how quickly or how slowly you turn them. The text displays keep you oriented at all times and even offer help files when you need them.

SAC stands for Software Assigned Controller. Like any control surface, the degree of control depends on the software support provided by the developer. You don't have to bother with assigning functions to the SAC-2K's controls; everything it needs to know is built into your software and the SAC-2K's firmware.

If your life revolves around *Digital Performer* or *Pro Tools*, the SAC-2K is calling your name. You can use every slider, knob, and button to access literally hundreds of parameters. You can even access dozens of plug-in parameters to control effects processors and virtual instruments. Music applications from Steinberg, Emagic, Propellerhead, Native Instruments, CreamWare, and others are also supported, with many more in the works. The release of the SAC-8X should provide further expandability through the SAC-2K's USB port. The SAC-2K is a product to watch, and it's unquestionably the best new control surface of the past year.

Digital Audio Sequencer

CAKEWALK
SONAR XL 1.02 (WIN; $739)

MARK OF THE UNICORN
DIGITAL PERFORMER 3.01 (MAC; $795)

Picking a winner in this category is never easy. As we did two years ago, we gnashed our teeth and wandered around in circles, mumbling to ourselves, but we couldn't seem to settle on a clear winner. Our Windows-savvy editors wanted to honor Cakewalk's *Sonar XL* 1.02 for its redesigned user interface and powerful new features. The Mac enthusiasts lined up solidly behind Mark of the Unicorn's (MOTU's) *Digital Performer* 3.01.
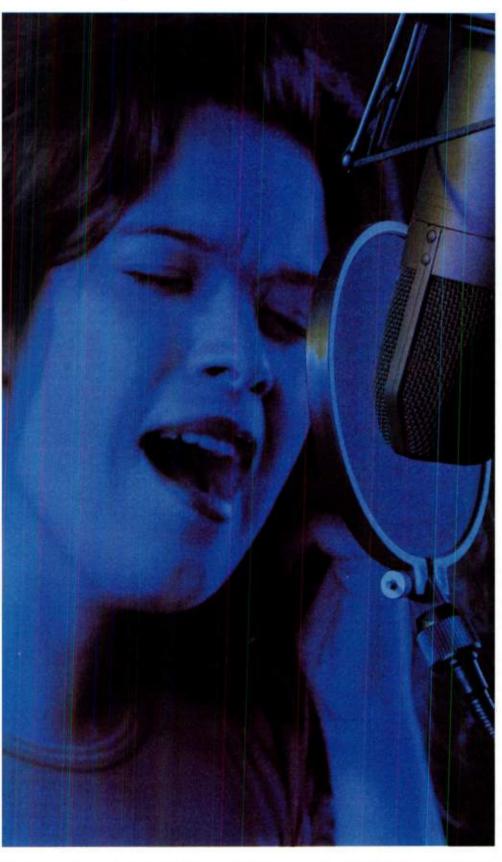
With our house divided sharply along party lines, we avoided bloodshed by declaring a tie. After all, both programs are powerful enough for professional work, offer an unlimited number of tracks, support nondestructive editing, and let you view and edit audio and MIDI data in the same window. It seemed only fair that we share our praise with both camps.

Although *Sonar XL* owes much to its predecessor, *Pro Audio* 9, it represents a significant departure from the past in several important areas. Under-the-hood enhancements include support for Microsoft's new WDM technology, which offers dramatically

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PEAVEY

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lower latency than the old MME drivers. With WDM drivers, you can use real-time effects while monitoring live input and apply real-time effects during recording. You also can play software synths from an external controller without undue latency.

Another significant development is *Sonar XL*'s support for DirectX 8 plug-ins and the new DX Instruments (DXi) format. They let you incorporate a wide range of powerful processors and software synths into *Sonar XL*, and with *Sonar*'s new automation features, you can automate processor parameters and synth settings to create exotic morphing effects and sounds. To help you take advantage of all that sonic potential, *Sonar XL* includes several DX Instruments, including Audio Simulation's *DreamStation*, Roland's *Virtual Sound Canvas*, and Applied Acoustics' *Tassman* (an Editors' Choice award winner last year).

For processing audio, Cakewalk throws in its *Audio FX* 1 and 2 plug-in collections, five excellent new DirectX effects from Power Technology, and *ReValver* from Alien Connections. Those make quite a bundle, but *Sonar XL* has much more to offer. The program's new Loop Construction and Loop Explorer views combined with its Groove Clips feature may give Sonic Foundry's *Acid* a run for its money. Groove Clips can match the playback tempo and pitch of each loop in a project, and *Sonar XL* can even import *Acid* loops for maximum compatibility. The Groove Clips feature allows *Sonar XL* users to create complex loop-based projects without leaving their primary sequencer—very cool. *Sonar XL* is an important new tool for Windows-based music production and deserves to be recognized for its clean and friendly design and its collection of powerful tools.

Also deserving to be highlighted is MOTU's *Digital Performer*, a true titan among sequencers. Its well-crafted user interface and mile-long list of features make it nearly undefeatable in a competition. After it tied with Emagic's *Logic Audio* two years ago and then won last year's Editors' Choice award outright by pum-

meling us with myriad new features in version 2.7, we weren't prepared for yet another major upgrade. We were stunned when we got our first look at *Digital Performer* 3.01 (for a review, see p. 120). For starters, the program has undergone a complete face-lift and sports several new windows, slide-out drawers in the Control panel, and an overall look and feel that coordinates nicely with Apple's new Aqua user interface for Mac OS X.

The changes, however, go far deeper than a jazzy new appearance. For example, *Digital Performer* is now optimized to get the most from dual-processor G4 Macs, providing blazing speed and newfound power when working on complex mixes and surround-sound projects. *Digital Performer* 3.01 embraces surround sound in a big way, with lots of versatile tools and support for a broad spectrum of formats. Among other things, the program includes four new surround-panning plug-ins offering sophisticated control of sound placement. Speaking of plug-ins, *Digital Performer* now includes a whopping 35 great-looking, great-sounding, professional-level plug-ins for tackling just almost any processing task imaginable.

Digital Performer 3.01's new Sequence Editor window is a significant enhancement. It lets you view and edit audio and MIDI tracks in the same window, and if you're using QuickTime movies for scoring or other post-production work, you can view the individual frames along the timeline in the new Movie track as well as in a separate dedicated Movie window. Dragging audio clips, automation points, and other editing elements scrubs the onscreen movie, making frame-accurate placement a snap.

The sequencer boasts far more high-end features than we can cover here. Nevertheless, its automation capabilities are awesome; it can import and export *Pro Tools* projects via OMF; its redesigned Graphic Editor window can simultaneously display data from multiple tracks in different colors; and its new Audio Bundles window lets you quickly reconfigure hardware assignments as you change projects. *Digital Performer* 3.01's documentation is among the best in the industry. We could go on, but you get the point. We just had to give this amazing program an award—for the third consecutive year.

DAW/Audio Interface

MARK OF THE UNICORN
828 (MAC/WIN; $795)

The world is awash in multichannel digital-audio interfaces for desktop music systems. Until recently, however, all of the current multichannel systems—whether for Mac or PC—centered around a PCI card. We saw many excellent audio interfaces with top-notch specs this year, but one product stood noticeably apart from the crowd.

The 828 from MOTU is the first computer-based,

Now you can sound as good "Live" as you do in the shower.

You're standing in the shower, soaking wet, when all of a sudden inspiration strikes, and before you know it you're belting out a tune at the top of your voice. You sound great, and you know you could be a big star if you could only sound this way "Live."

Well thanks to DigiTech, now you can. Because besides being a very cool mic pre, the Vocal300 from DigiTech is equipped with 26 fully programmable studio quality effects (you can use up to 7 effects at once), along with a built-in expression pedal and footswitches that let you change your sound on the fly. There's even a collection of "character" voices that will transform even the silkiest voice into a demonic fiend or a space-crazed alien.

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multichannel hard-disk recording system to use a high-speed FireWire interface rather than an audio card or pokey USB interface, thereby bringing high-speed multichannel recording to Macs and PCs without using an expansion slot. Whether you have a desktop computer crammed with PCI cards, a FireWire-equipped laptop, or a slotless tabletop computer, you can create a quiet, compact, fanless system that's a cinch to set up.

Of course, there is no glory in being the first to do something if the product you introduce is not up to snuff. Fortunately, the 828 delivers tremendously in spite of its modest price tag. The unit offers eight channels of analog I/O, eight channels of ADAT Lightpipe I/O, coax or optical stereo S/PDIF I/O, ADAT sync, zero-latency monitoring, and two mic preamps with switchable phantom power. The balanced ¼-inch analog jacks use 24-bit converters with a 105 dB dynamic range. Inputs 1 and 2 use Neutrik XLR/TRS combo connectors, so you can plug mics as well as guitars and sound modules in to the jacks. When we reviewed the 828 in November 2001, we found the sound quality to be excellent and the mic preamps to be quiet enough for pro-level work.

We are pleased that MOTU includes the same *Audio-Desk* Mac software that comes with the company's popular 2408 PCI-based system. *AudioDesk* is essentially the audio portion of *Digital Performer* 2.4, and it turns the 828 into a formidable recording and editing system. The software includes MOTU's PureDSP time-stretching and pitch-shifting technology as well as dozens of powerful 32-bit real-time effects plug-ins.

From its front-panel headphone jack, volume controls, trim controls, and banks of LEDs to its rear panel bristling with I/O options, the card-free MOTU 828 would be a winner all by itself. Add the *AudioDesk* software to the mix, and you have a solid, affordable, digital audio workstation that is truly hard to beat.

Dynamics Processor (hardware)

DRAWMER

DS501 POWER GATE ($550)

Since when is using a noise gate fun? Since Drawmer released the DS501 Power Gate. Typically, gating is regarded as a perfunctory chore, something that must be done to reduce leakage and maintain clarity in a mix. But the new dual-channel Power Gate may change that

perception, thanks to its sophisticated design, precision control, and wealth of features. As our surprised reviewer put it, "With the 501, gating is no longer a dreary chore—it's actually creative and fun." (For the review, see p. 134.)

In addition to the usual envelope-control parameters found on full-featured gates—threshold, attack, hold, decay, and range—Drawmer has outfitted the Power Gate with several useful new features. Most noteworthy is the innovative Peak Punch, a circuit that allows for tunable transient boosting. Designed primarily for drum and percussion gating, Peak Punch provides a 10 ms gain boost each time the gate opens, reinforcing the initial transient of the gated audio—a real problem-solver for wimpy or ill-defined drum hits. Perhaps more useful still is the unit's Key Filter. In addition to permitting standard gate triggering via a key-input source, this section also provides variable low- and high-frequency shelving filters, allowing for "frequency-conscious" gating without the need for an outboard equalizer. How's that for slick?

As great as the Power Gate is for drum-gating duties, we are duly impressed by how well it performs on less percussive sources such as vocals, guitar, saxophone, and synthesizer. Given the depth of control granted by the Power Gate's extensive feature set, the unit's versatility and creative potential should have come as no surprise. The same can be said of other qualities that make it our favorite dynamics processor of the year: quiet operation, minimal coloration, ample headroom, superfast attack times, and noiseless bypass switching.

Effects Processor, Analog (hardware)

DACS

FREQUE II ($1,400)

The past year was a good one for analog effects processors, including Demeter's smooth-sounding RV-1 spring reverb and the terminally hip Sherman Filterbank II. However, when it comes to mangling your music in delightful ways, the DACS Freque II takes the prize. The Freque II puts a pair of ring modulators in a 2U rackmount box. The ring modulators can be operated independently—each processor has

its own program and carrier inputs and dedicated controls—or linked into Dual-Mono mode, so that both processors are controlled by one set of knobs.

Engage the FM On switch, and oscillator 1 is routed to the CV input of oscillator 2 for instant frequency modulation. A dedicated FM Depth control lets you add modulation to taste. For real tweaking satisfaction, push the Freque button: it transforms the processor into a frequency shifter, which gives you the

sum and difference of tones at separate outputs. The amount of shift is controlled by oscillator 2 and can be changed manually or with a control voltage. The frequency shifter gives you a range of effects, from subtle stereo processing to wild timbral shaping.

Compared with the original Freque, DACS has brought Freque II up to a pro level, adding trim controls to the four audio inputs, a CV input jack and oscillator output jack for each ring modulator, separate level controls for each output, and balanced TRS jacks throughout.

To some, ring modulation and frequency shifting may seem like exotic effects. But the simplicity with which you can shape sound using the Freque II, whether subtly or extremely, makes it a joy to use—not to mention a colorful addition to your effects rack.

Effects Processor, Digital (hardware)

LEXICON

MPX 200 ($399)

Nearly all inexpensive effects processors offer compromised control options and effects quality to keep costs down. Lexicon's MPX 200 is an exception. It delivers high-quality effects, a hefty amount of real-time and MIDI control, digital I/O, and an independent stereo compressor.

The single-rackspace processor is useful for both live and studio applications. It has a high-impedance input, so you can plug a guitar or bass directly in to the unit. Its coaxial S/PDIF I/O lets you put it to work as

an analog-to-digital converter. As a component in a P.A. system, you can send a processed stereo signal to a monitor mix while sending a dry output to a DAT, for example, through the unit's digital outputs.

The MPX 200 furnishes flexible signal flow settings: Parallel gives you dual stereo I/O, Cascade feeds one effect into the other, Mono Split offers two mono inputs that feed two separate stereo effects, and Dual Mono sends a single mono effect to each output. At the front of the signal chain is a digital stereo compressor with ratio, threshold, attack-time, and release-time parameters. The compressors offer transparent sound with minimal pumping, albeit without the characteristic warmth and grit of analog compressors.

Time-based effects, such as delay, offer several tempo-synchronization options. You can send an analog signal from your instrument to set the tempo, use the front-panel Tap button or a footswitch, or lock the effect to incoming MIDI Clock. Furthermore, the MPX 200 supports real-time MIDI control of parameters using Pitch Bend, Aftertouch, and a variety of Control Change messages. For example, you can send MIDI messages to adjust a room reverb's "liveness" parameters or change the rate of a rotary effect.

The MPX 200's greatest strength is its sound. In addition to its array of sparkling choruses and flangers, pitch shifters, rotary simulators, and more, the unit provides smooth, clean reverbs that sound remarkably close to Lexicon's legendary high-end effects processors. That combination of pristine sound, flexible signal routing, real-time MIDI control, and 24-bit digital I/O is hard to top, especially for less than $400.

Loop Sequencer

SONIC FOUNDRY

ACID PRO 3.0 (WIN; $399)
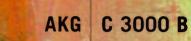
When you think of loop-assembly tools for the PC, one product comes to mind before all others: Sonic Foundry's Acid. Best known for its real-time tempo-matching and pitch-shifting features, Acid Pro, now in version 3.0, allows you to incorporate MIDI data into your projects and "paint" it onto the work area with ease. What's more, you can add a single video track to the Acid work space, making the program an excellent audio-for-video platform.

Acid 3.0's new Chopper tool lets you easily create complex rhythmic patterns using small segments of your loops. Also new are more than a dozen DirectX plug-ins that give you even more ways to tweak your tunes. If you are ever running short of loops, go to Sonic Foundry's Web page and pick up any of the numerous loop libraries, all of which have been fine-tuned to make your Acid experience more enjoyable.

The program has a number of functions that make interaction with the "outside" world easy. You can rip

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**ROLAND
VS-2480**

The C 3000 B is the reference
standard for Roland's V-Studio
workstations with COSM
technology

AKG

RAISE THE LEVEL

and burn CDs, transfer files to your hardware sampler, link to Sonic Foundry's Acid Planet Web site (www.acidplanet.com) to upload your files, and more. You'll

even find a copy of *Sound Forge XP* on hand, which allows you to tweak or mangle your samples without quitting the program. Talk about integration!

We've been hooked on *Acid* since its inception. If you haven't had the pleasure, there's no time like the present to give it a try.

Microphone (over $1,000)

BALTIC LATVIAN UNIVERSAL ELECTRONICS
KIWI ($2,299)

Once in a great while, a product comes along that sets new standards for excellence in its class. The Kiwi microphone from Baltic Latvian Universal Electronics (BLUE) is such an item. This is the third consecutive year BLUE has captured an Editors' Choice award: the Dragonfly won in the over-$1,000 category last year, and the Blueberry was our pick for large-diaphragm condenser in 2000. Clearly, BLUE is on to something, and we know what it is: the company is building a reputation not on cost-effective mass production, copies of classic vintage designs, or cheap transducer imports from abroad but on distinctive, first-rate microphones that are solidly constructed from the finest components.

Admittedly, **EM**'s emphasis on value rarely permits reviews of mics as pricey as the Kiwi. Considering what you get for your dollars, however, the Kiwi is a bargain. After all, it is BLUE's top-of-the-line solid-state microphone. Treasure that it is, the Kiwi ships nestled in sumptuous purple velvet inside a gorgeously crafted cherry-wood box. A high-quality, spider-type elastic shockmount is included—something you pay handsomely for from other mic makers.

Like all BLUE mics, the Kiwi employs Class A discrete circuitry (no ICs) and a hand-built, hand-tuned capsule that is the main agent of the mic's high-quality

sound reproduction. That stands in sharp contrast to designs that use EQ to "improve" an initially substandard sound. In addition, the Kiwi uses transformerless output circuitry and a dual-diaphragm capsule that allows for nine polar patterns, including cardioid, omnidirectional, and figure-8. The range of patterns, which is unprecedented in a solid-state condenser, greatly extends the Kiwi's versatility, allowing users to fine-tune the frequency response and find the ideal character of sound pickup for a given source.

Without a doubt, the Kiwi sounds wonderful: with open, silky highs that never sound brittle or harsh, it is exceedingly natural and transparent overall, yet surprisingly mellow for a FET design. The images the Kiwi captures have remarkable depth and solidity, a quality only the best transducers seem capable of producing. Not surprisingly, the Kiwi is also supremely quiet, has a huge dynamic range, and exhibits superb transient response. We especially like it on vocals, acoustic guitars, piano, organ, Dobro, dulcimer, mandolin, drums, and all sorts of percussion. Simply put, the Kiwi is both a workhorse mic and a work of art.

Microphone (under $1,000)

RØDE MICROPHONES
NTK ($999)

Røde has long been on the front lines of the microphone bang-for-buck shift—you know, the one that brought quality large-diaphragm condenser mics within reach of the personal-studio buyer. This is Røde's second **EM** Editors' Choice award: the company also bagged one in 1998 for the NT1 (originally $499, now $349), a microphone that its maker recently proclaimed "the world's No. 1-selling studio mic." Although we can't vouch for that claim, we can make one we deem more important: Røde's latest creation, the NTK, is the finest large-diaphragm tube condenser for less than a grand (for a review, see p. 130).

Granted, as tube condensers go, the NTK is pretty bare-bones; it ships in a cardboard case with a zipper pouch, a power supply, a 30-foot cable, and a simple clip. The mic, too, is minimalist, with no pad or filter and only a cardioid polar pattern. What you do get, though, is rock solid (the NTK weighs in at nearly two pounds) and lovely, thanks to a beautiful satin-nickel finish.

But what really turns our heads is its sound. As our reviewer notes, "The NTK was designed to be musically pleasing rather than clinically accurate." This mic has a big rich sound with solid, unhyped lows and mids as well as a presence boost that peaks at 5 and 12 kHz. The result is a bright, in-your-face sound, yet the brightness is pleasantly tempered by

the twin-triode 6922 tube (or *valve*), which imparts a subtle sheen and roundness of tone.

Something else is going on, too: personality. Rather than a linear response, the NTK's character changes as you work the capsule closer, seeming to blossom with attitude. That makes it a great vocal mic, of course, and excellent on other sources, including piano and acoustic guitar (steel and nylon string). Moreover, the NTK is remarkably quiet for a tube mic, has a substantial dynamic range, and can handle inordinate levels of sound pressure. Considering that all those qualities come together in a tube mic for less than $1,000—well, we think that's one heck of an achievement.

MIDI Instrument Controller (nonkeyboard)

STARR LABS

Z6 AND Z6-S ($1,995; $2,095)

Harvey Starr's guitar-style instruments offer virtually unparalleled MIDI control. The Z6 provides six rubber strum bars, whereas the Z6-S sports a bridge with a set of 6-inch-long string triggers that offer a more guitaristic approach. However, the Z6 and Z6-S go well beyond mere triggering of MIDI note data.

The fingerboard buttons can send Note-On messages, enabling two-handed neck-tapping techniques. The pressure-sensitive buttons can send any MIDI continuous controller message; in fact, you can assign zones to areas of the neck to send different messages for each zone or assign zones to different MIDI channels. Zones can even send entire sequences for self-accompaniment; zoning allows alternate note assignments, so players are not locked into conventional guitar tunings. Thirty-two user locations let you store zone and MIDI control assignments as patches, so you can quickly switch the instrument from one complex setup to another.

The Z6 and Z-6S provide an abundance of ergonomically arranged options for creating stimulating music. In addition to the strings, fingerboard, and strum plate, each instrument has an assignable joystick with which you can send different MIDI messages for the four directions. If that's not enough, the joystick's firing button can send another independent message, but that's not all: you can add an assignable NeckSensor pressure strip, six independent expression pads that sit on the sides of the Z6-S bridge, and a proprietary breath controller. Dedicated MIDI controllers should go beyond the pale to provide tools for expressive music. The Z6 and Z6-S offer the best of guitar and keyboard technologies to wring every last ounce of expression from your MIDI gear.

Modular Hard-Disk Recorder

MACKIE

HDR24/96 ($4,999)

Two contenders battled for the top spot in the Modular Hard-Disk Recorder category, and it was a close call. The iZ Radar 24 put up a good fight, due in large part to its stability, sound quality, and ease of operation. Notwithstanding, in addition to having similar attributes, this year's winner, the Mackie HDR24/96, adds a DAW-like user interface to its feature collection.

The HDR24/96 is a solid and robust recording system that is about as easy to use as they get. The unit operates much like a tape machine, and you can configure it from the front panel or from within the software graphical user interface. Herein lies one of the HDR24/96's nicest amenities: connect a computer keyboard, mouse, and monitor to the HDR24/96, and you can take the system beyond basic tracking chores.

Although the DAW-style editing features are not as deep as those in a computer-based DAW, the HDR24/96's software supplies the essentials for moving and editing audio files. Mackie chose a

It's easy to make it look different.
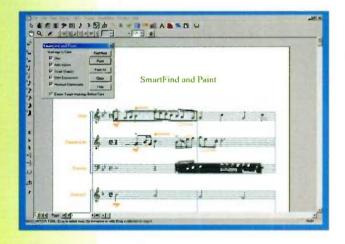
But difficult to make it sound better.

Model 2029B

Model 1031A

Model 1037B

favorable amount of redundancy between the software and hardware front-panel functions, so you can easily jump between interfaces.

You'll also appreciate the variety of I/O options, including analog, AES/EBU, TDIF, and Lightpipe. Not surprising, the HDR24/96 integrates well with another important Mackie product, the Digital 8-Bus mixer. The HDR24/96 also can act as an FTP server for easy file transfers to your computer while the device is connected to a Mac or PC through the rear panel's 100Base-T Ethernet port.

The HDR24/96's stability, excellent audio quality, full feature set, and user-friendliness make it worthy of this year's award. To quote Larry the O's HDR24/96 review, "It's a winner."

Notation Software

CODA MUSIC TECHNOLOGY
FINALE 2002 (MAC/WIN; $545)

When Coda Music Technology introduced *Finale 2000*, we were mighty impressed by its new Setup Wizard, dozens of templates, sophisticated Maestro font, and great documentation. Just as we were bestowing the 2001 Editors' Choice award on *Finale 2000*, however, Coda released *Finale 2001*, which added Internet-publishing capabilities, music-scanning technology, and the ability to transcribe melodies from a mic input.

At that point, you would think Coda would rest on its laurels and catch its breath. Instead, the company promptly introduced *Finale 2002*, a significant upgrade with an amazing array of exciting new features. *Finale 2002* provides enhanced Simple Entry and Speedy Entry modes and a savvier Selection Tool for quicker and more intuitive note entry and score editing. Contextual menus make it easy to set time and key signatures on the spot and offer immediate control of hairpins, expression marks, slurs, and other Smart Shapes objects, which now include more grab handles for better positioning. Slurs are smarter and more beautiful than ever, with options for automatically avoiding collisions with stems, beams, accidentals, and other markings.

Finale 2002's powerful new editing and note-entry features are impressive, but what really surprises us is the program's evolution far beyond the usual music-notation capabilities and emergence into the world of music production. For example, band and orchestra teachers will love the new Exercise Wizard. Just choose from a long list of exercise types, specify the instrumental ensemble and a few other parameters, and *Finale 2002* generates a complete exercise lesson with individual parts for each player. But that's just the beginning.

The software also includes ten Composer's Assistant plug-ins—algorithmic music-creation and manipulation tools developed in Paris by IRCAM. The plug-ins can generate smooth transitions between chords or from one melody to another. They can find new placement options for chords, produce a series of chords with increasing complexity and texture, generate four-part triadic harmonies, or create as many as six staves of percussion to accompany a score.

Finale 2002's other new plug-ins help you align markings, create tremolos and string harmonics, and add playback to trills and glissandi. One of our favorites plug-ins is *Band-in-a-Box Auto-Harmonizer*. Based on the popular auto-accompaniment program from PG Music, it automatically generates two- to six-part harmonies in a range of styles, from Barbershop to Super Sax, for any melody with or without chord symbols.

Finale 2002 boasts far more new features than we can list here, and it has once again amazed us with its scope of notation options, powerful editing tools, and creative music-generating features that push the music-notation-software envelope. Despite some promising new competitors this year, *Finale 2002* has triumphed again as the most noteworthy music-scoring program.

Monitor Speaker (close-field)

TRUTH AUDIO
TA-1P ($999 PAIR)

In recent years, the spotlight has been on powered monitors, mostly made by large well-known manufacturers, but the close-field monitor that made us sit up and take notice this year is a passive design from a small new company called Truth Audio. "But wait, aren't passive monitors passé?" you ask. Not when they sound this good; we were impressed by the TA-1Ps from the moment we laid ears (and eyes) on them.

The TA-1P is distinctive for employing two 5-inch woofers, positioned slightly below and on either side of a 1-inch cloth-dome tweeter. Don't be fooled, though—those two little woofers couple with the TA-1P's rear-ported cabinet to produce tight, focused low end down to 50 Hz. Highs and mids are faithfully represented, too. Indeed, the TA-1P's frequency response fluctuates only ±2 dB from 51 Hz to

22 kHz—a real achievement in a passive monitor at this price.

In the end, sound and performance count, not specs. Reviewer Rob Shrock, a demanding veteran with years of experience evaluating close-field monitors, loved mixing on the TA-1Ps. He was especially impressed by their exceptional low end, excellent imaging, and uncanny ability to present an even-sounding frequency response at different playback levels. Although we detected a slight bump in the low-midrange response, overall the sound was neutral, natural, and wonderfully nonfatiguing, even after hours of close monitoring. Most important, TA-1P mixes translated exceptionally well to other playback systems. If you're ready to hear the truth about your mixes, Truth Audio is a sensible place to go.

Portable Digital Studio

ROLAND

VS-2480 ($4,495)

Roland's VS-2480 portable digital studio offers many of the graphical editing benefits of a computer-based digital audio workstation. With its support for an ASCII keyboard and mouse, you can drag and drop track regions as well as select regions for copy, paste, move, and more. Connecting the unit to a VGA monitor provides large-screen views that resemble those of a software digital audio sequencer. For instance, a Track View window provides track numbers and names on the left and a playlist showing recorded regions for each track on the right. The VGA display also provides a view of track channel strips. Dialing in an EQ setting opens up a display that graphically illustrates changes to the selected frequency band in real time.

The mixer section sports 17 motorized faders, and the unit records all moves for posterity. Hopefully, you'll never need all 999 levels of undo, but they're available if you want them. You also get 384 virtual tracks for alternate takes. If you like to work with sample-construction kits, you can use the VS-2480's Phrase Pads, which let you sequence audio by triggering audio regions. You can trigger imported WAV files from CD or trigger selected re-

gions directly from the unit's internal hard drive.

You can record as many as 16 tracks at a time with the recorder's 8 phantom-power XLR jacks and 16 balanced TRS inputs. The VS-2480 gives you a maximum 24 tracks and as much as 24-bit, 96 kHz audio. Internal 64-bit resolution ensures plenty of processing headroom, and the recorder includes two great-sounding stereo multi-effects processors and slots for six more.

The VS-2480 sounds terrific and is capable of producing professional sounding results without outside help. Nonetheless, the unit offers SMPTE and Word Clock input as well as dual R-Bus ports for expanding the digital and analog I/O.

The competition was tough, but with its great sound, expandability, connectivity, and 24-bit, 96 kHz capabilities, the VS-2480 found its way to the head of the pack.

Preamp (mic)

GRACE DESIGN

MODEL 101 ($699)

We editors normally balk at manufacturer ad copy, but no one rolled their eyes or muttered "Yeah, right" when Grace Design described its Model 101 as the "ultimate microphone preamplifier for under $1,000." Why not? Well, for one thing, as a major player in the small but fiercely competitive world of boutique mic preamps, Grace Design has no reason to make exaggerated claims. The solid-state, Class A Model 101 is basically a mono version of the company's highly acclaimed Model 801; as long as Grace was able to fit the same amplifier architecture into a 1-channel box, we figured, it surely had something worth chortling about.

We didn't leave it at that, of course. Rather, we sent the Model 101 to Myles Boisen, one of our toughest reviewers. After putting the unit through a virtual obstacle course of performance tests, including comparisons to legendary preamps cost-

ing three and four times as much, he gave the unit top marks across the board; only in the features rating did the device fall a half-point shy of perfection. Boisen pronounced the 101 "predictably neutral in character and impressively close to the technical ideal of a wire with gain." He was especially struck by the 101's seemingly

endless high end, which he described as "extending into realms only dogs and test gear could appreciate."

But the 101 doesn't stop there: it also provides a ¼-inch jack on the front panel for direct injection duties. Here its performance is every bit as uncompromising as on mic signals; the clarity, articulation, and wonderfully defined low end it delivered on electric guitar and bass are stunning. Naturally, considering its pedigree, the 101 is exceptionally quiet (the signal-to-noise ratio is greater than 92 dB), and it provides plenty of gain, making it a great choice when using ribbon mics or recording quiet sources. As if all that weren't enough, the Model 101 is also ruggedly constructed and beautiful to behold.

Sample Library

HARPS.COM
GARRITAN ORCHESTRAL STRINGS
(GIGASAMPLER; $1,399)

EM looked at more than 75 sample CDs and CD collections this year, and many new offerings were impressive. However, picking a single winner from a year's worth of widely diverse contenders invariably leaves us feeling dazed and confused, so we generally don't give an award for best sample library. This year, however, we discovered a CD collection that truly blew us away. Harps.com's *Garritan Orchestral Strings*, created by Gary Garritan, is exceptional not only in the depth and breadth of its content but also in the thorough and thoughtful way in which it is presented.

Garritan Orchestral Strings comes on 16 CD-ROMs that focus exclusively on the string sections of the orchestra. The 24-bit recordings were made at New York's Lincoln Center and were performed by top-notch symphonic players. Each section offers a dizzying array of articulations, bowing techniques, and special effects sampled chromatically at multiple dynamic levels and

without loops. Clever controller routing maximizes the expressive potential of the patches, which were derived from more than 8,500 individual samples. For example, you can increase a sound's warmth by sliding a fader, you can cross-fade from nonvibrato to vibrato samples with the Mod wheel, or you can quickly change bowing techniques with key switches. The overall sound is lush and warm.

But the library itself is not the whole story. Garritan really pushes the sample-library envelope by adding several special benefits to the package. To provide capabilities beyond those in *GigaStudio,* for example, Garritan created *MaestroTools,* a program that performs tricks such as automatically alternating between up-bow and down-bow samples as you play. Moreover, the *Garritan Orchestral Strings* documentation is exemplary; it goes far beyond what we've come to expect with sample libraries. Aside from a complete and detailed listing of presets and controller routings, the 120-page hard-copy owner's manual includes sections about using the library effectively, working with reverb and panning, elements of string writing, instrument ranges, and the layout of the orchestra as well as a glossary of string terminology, a 6-page bibliography, and a great deal more.

But that's not all. The Garritan library is updateable, so you can add new capabilities as they become available. As of this writing, the first update disc has already been sent to registered owners, and more discs are planned. Finally, Garritan provides a helpful and congenial online forum through which his library users can trade tips, help each other out, and generally function as a close-knit community.

Garritan Orchestral Strings is an exceptional product, combining a beautifully rich-sounding string library, supplementary software, intelligent

controller programming, unparalleled documentation, open-ended evolving content, and membership in an online community. This sample library clearly stands head and shoulders above the crowd, and that is a very special achievement.

Sampler (software)

NICK WHITEHURST
SAMPLERCHAN 1.3 (WIN; $69)

SPEEDSOFT/MAZ
VIRTUAL SAMPLER 2.56 (WIN; $75)

One of the great joys of working at **EM** is having the chance to spread the word about software from small companies that might not otherwise make it into the big time. Speedsoft/Maz's *Virtual Sampler* and Nick Whitehurst's *SamplerChan* are great tools for tweaking and playing samples on a PC, and both offer a number of features that move them well beyond the normal sampler model.

Virtual Sampler, which runs standalone and as a DirectX or VST Instrument, offers an economical interface in which you toggle among the various work areas. At the top of every screen are the tools for file management and program configuration, and along the bottom, you'll find a virtual keyboard to trigger the sounds you're working on. You need a powerful computer to max out the program's 64 available polyphonic notes, but at least you can't complain that *Virtual Sampler* is holding you back.

Virtual Sampler can load and save files in numerous formats, including WAV, AIFF, TerraTec's TTI instruments, LM4 drum kits, Akai, and SoundFont2. Unlike most software samplers, it streams samples from disk, so no file-size limit is set. Among its other unusual features are the ability to process the audio output of VST Instruments through its effects section, a customizable skin interface, and the inclusion of both a DX7 clone and a classic analog-style software synth. You can also apply VST plug-in effects to your sounds, which gives you a nearly unlimited library of sound-manipulation options.

Cakewalk has just become the exclusive distributor of the DXi version of *Virtual Sampler*, so you should have no trouble getting updates and patches when needed. A random sampling of the hundreds of messages from users at the *Virtual Sampler* Forum suggests that this is one very popular package.

Nick Whitehurst's *SamplerChan*, a VST Instrument for the PC, brings its own strong hand to the table and includes many tools designed to aid in composition and real-time performance. For example, it can process an incoming MIDI note and use it to trigger a long pitch sequence you create in the program's Rhythm Section, or automatically extract the beats from a groove. Setting up a multisampled patch couldn't be easier: just drag your sample files right onto the program's interface. Creating a multitimbral patch is just as simple.

Like *Virtual Sampler*, many of *SamplerChan*'s features can be controlled via incoming MIDI data, which makes for a lively and spontaneous sampling experience. The program offers as many as 128 polyphonic notes and can route its output on a total of eight separate channels. It offers extensive matrix-modulation options that allow you, for example, to modulate the level of effects sends in your patch. You'll find lots of help for looping your samples in the integrated wave editor, and the excellent manual is a great resource for getting a handle on the program's features.

SamplerChan's Performer is a powerful inclusion that you won't find elsewhere. It acts like a virtual joystick controller and lets you manipulate any number of parameter settings in real time. The Performer is just one of many features that make *SamplerChan* a great tool for live performance.

Signal-Processing Plug-In

AUDIO EASE
ALTIVERB (MAC; $495)

This category always is a tough one. We could have happily given the award to Antares Systems' fabulously popular *AutoTune* 3, Metric Halo's eminently useful *Channel Strip*, or Native Instruments' innovative *Spektral Delay* (which won the award as a stand-alone signal-processing program). But we ended up with Audio Ease's groundbreaking yet easy-to-use

Altiverb, a MAS reverb plug-in for the Mac that raises the bar for this type of processor.

The significance of *Altiverb* is that it uses impulse-response technology to re-create the reverb characteristics of an acoustic space in real time on a stock Mac G4. Using impulse-response technology for that task is nothing new, but to do it in real time is expensive and CPU intensive, and dedicated outboard processors with that capability can cost thousands of dollars. So when Audio Ease successfully brought that technology to the desktop studio, it immediately got our attention.

For those unfamiliar with the technology, here's how an impulse-response-based reverb is created: First, an acoustic space is *sampled* by recording an impulse (often a shot from a starter's pistol) in the room. Next, the impulse is subtracted from the recording of the acoustic space through deconvolution. The result is a time-variant filter that mathematically represents the characteristics of the room. The impulse-response method is a highly accurate way to re-create an acoustic space, but you can see why it takes a lot of computing power.

Altiverb offers as much as three minutes of reverb based on the room of your choice, and it can be configured for one, two, and four channels. Out of the box, the plug-in includes samples of eight environments, including the famed Amsterdam Concertgebouw. There are more than two dozen variations of this building alone, based on the quantity and type of microphones used in the sampling sessions and their placement in the room. Other spaces include St. Joseph Church and Vredenburg Concert Hall in Utrecht, schoolrooms, and the interiors of various cars and vans. *Altiverb* also lets you sample your own acoustic spaces.

Remarkably, *Altiverb* is easy to use. Simply choose the environment you want and tweak the four controls: Predelay, Decay Time, Wet, and Dry. That's it. The result is a highly realistic reverb.

To do the massive amount of convolution re-

quired, Audio Ease tapped into the Motorola PowerPC G4's powerful AltiVec Velocity Engine, so you must have a G4-based Mac, and even then the CPU demands are formidable. Compared with the price of the hardware sampled-acoustics processors on the market, you can buy *Altiverb* and the fastest computer and still have money left over.

Signal-Processing Plug-In Bundle

WAVES
GOLD NATIVE 3.0 (MAC/WIN; $1,300)

The latest version of Gold Native is more powerful than ever, rivaling the capabilities of the almost 20 hardware devices it emulates, including tools for tracking, tweaking, sweetening, mangling, and mastering live and recorded audio. The package currently supports a variety of plug-in formats, including VST, MAS, RTAS, and AudioSuite for Macintosh and VST, DirectX, and RTAS for Windows.

Although version 3.0 takes full advantage of the processing speed made possible by the latest computers, Waves recently updated its plug-ins to vastly reduce the processing power they require, giving the ability to run more simultaneous plug-ins. In addition to new, improved, and consistent user interfaces and increased parameter resolution, you get automated parameter control, extensive documentation, hundreds of factory setups, and support for 88.2 and 96 kHz audio. Another welcome improvement is the challenge-and-response copy-protection scheme that has replaced the dongle required by earlier versions.

Gold Native 3.0 also introduces two new plug-ins: *C4 Multiband Parametric Processor*, which offers 4-band compression, expansion, limiting, and EQ, and *Renaissance Reverberator*, which emulates classic hardware and is certainly the best-sounding native reverb that Waves has ever offered. Add to

those the established plug-ins *UltraPitch, Enigma, MondoMod, MaxxBass,* and *PAZ Psychoacoustic Analyzer,* and you have a top-notch bundle that is virtually a must-have for serious computer-based musicians.

The Shortest Distance Between Inspiration Creation

Motif 6 and 7
Great Feeling 61-Note and 76-Note FS Synth Action Keyboards

Motif 8
Newly Designed 88-Note Weighted Balanced Hammer Action Keyboard

At Yamaha, we believe that making music shouldn't be work. So we designed the Motif Music Production Synthesizers with two goals in mind—inspiration and creation.

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

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

So if you want to be on the leading edge of music production, check out these websites for more info and then check out Motif at a Yamaha dealer near you.

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www.yamahasynth.com
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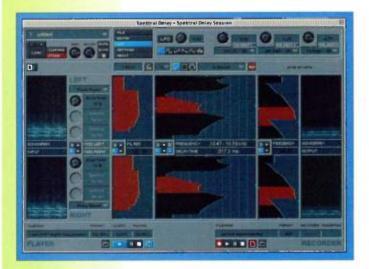
Signal-Processing Software (standalone)

NATIVE INSTRUMENTS
SPEKTRAL DELAY 1.0 (MAC/WIN; $299)

One look at Native Instruments' *Spektral Delay*, and you know that this software is unlike anything you've ever seen. Imagine having dozens of delay lines, each controlling a different frequency band, running at the same time. Now throw in real-time control of the amplitude, delay time, and feedback for each band, and you have yourself one awesome program.
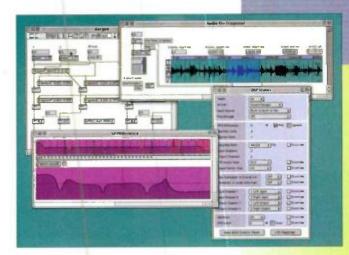
Spektral Delay analyzes a stereo audio file or real-time audio input and splits it into as many as 160 individual frequency bands per channel. You can modify each stereo channel independently, or you can link the two to make global adjustments. The graphic interface is easy to work with and fun to use, and you can spend hours tweaking the program's settings as a flood of amazing material pours forth from your speakers. Want to capture your sounds to disk? Turn on the Record function, and you're all set. (Just be sure you have a large hard drive!)

Spektral Delay has numerous tools for altering the delay parameters that control its output. You can apply various envelopes or repeating wave shapes to modulate the delay settings—for example, a sine shape on the feedback amount and a sawtooth on the delay time—and you can randomize one or more settings. You can also quantize, smooth, or average parameter values and use MIDI data to manipulate all parameters in real time. The preset configurations

make great starting points for your own explorations, enabling you to quickly generate a huge library of sounds to work with. A wide array of program settings help you get the best performance from your audio interface and computer CPU.

Revolutionary products such as *Spektral Delay* don't come around often, so grab a copy and start exploring new sonic domains.

Sound-Design Software

CYCLING '74
MSP 2.0 (MAC; $295)

Imagine a toolkit with dozens of modules for generating and processing sound in real time. Now add a graphical user interface that makes designing patches quick and intuitive, and you'll start to get a sense of what *MSP* is all about.

Put plainly, *MSP* does for audio what *Max* does for MIDI. (*Max* is required to run *MSP*; if you don't already have *Max*, you will need to buy the *Max/MSP* bundle for $495.) It offers nearly 200 audio objects that you can use in your patches, providing an almost limitless number of combinations. You can run your patches interactively so that a process begins only when it gets the specific trigger it requires, or you can build *MSP* patches that run completely unattended, perhaps generating random values for one or more objects. Modeling traditional synthesis methods, from FM to subtractive to granular, is simple, and you can use the included version of Cycling '74's *Pluggo* to turn your *MSP* patches into VST plug-ins.

Version 2.0 is a major upgrade that adds sample-accurate scheduling as well as polyphony and spectral-processing management. You can record and play audio files of as many as 24 channels with a maximum of 64 bits of resolution. The upgrade adds new objects, including one for designing filters graphically and another for monitoring the status of audio processing on your system; it provides support for even more file formats. Furthermore, the new audio-rate sequencer lets you integrate MIDI data into your patches with greater flexibility.

MSP supports most major Mac audio standards, including Sound Manager, ASIO, VST, Direct-Connect, and ReWire; it can also serve as a host for VST Instruments and effects. It ships with a large number of working examples, and its tutorials are second to none. Analyze, granulate, phase warp, or modulate your way to custom sounds with this all-purpose and powerful sound-design toolkit.

Sound Module

YAMAHA

DX200 ($630)

When you first consider its obvious features—a six-operator digital FM synthesizer with sampled bass and percussion sounds—the DX200 appears to be a DX7 with sampled rhythm-section sounds. But this impressive new desktop synth module is much more.

For starters, the DX200's work surface offers no less than 16 dedicated knobs for real-time control. You can smoothly move the synth from warm and fuzzy tones to clangorous and bell-like timbres. Real-time control of FM modulator amplitude offers smooth transitions between simple, sine-wave-like sounds to brighter, more distorted-sounding timbres. The resonant, multimode filter adds a cartload of warmth, depth, and animation to the characteristically static, brittle FM sound. You also get a knob for the filter's frequency cutoff and resonance amount, so you can dial up whistling, self-oscillating settings in an instant. Another knob lets you change the intensity of the filter's envelope from full-strength to completely inverted values. You can adjust ADSR envelope parameters in real time for both the filter and amplifier.

Unlike previous FM synths, the DX200 does a sur-

prisingly good job of emulating analog subtractive synth. Furthermore, the harmonic scope of a digital FM engine lets you venture into new sonic territory with a wider palette of sounds.

In addition to real-time control, the DX200's Free EG feature lets you capture and play back your sound-sculpting moves as a sequence of events. With four Free EGs to a patch, it's easy to create pads with sweeping, radically evolving timbral motion. Furthermore, you can engage the Free EG with the start of a DX200 pattern, a MIDI Start message, or a MIDI Note On message. The supplied software editor for Mac and Windows lets you create streams of

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KORG

Free EG events that can range from spiky and random to smooth and periodic.

The DX200 is compatible with the enormous library of six-operator FM sounds amassed since the DX7's heyday, and these sounds can seed tons of variations with a few tweaks of a knob. A modest complement of effects—including reverb, chorus, flange, delay, and distortion—add a final finish to the sounds. The unit's sample-playback engine adds a healthy assortment of exotic and natural drum and percussion sounds.

The sequencing component of the unit works well enough, but power users will want to connect the DX200 to an outboard sequencer. For example, an external sequencer will record the selection of new FM algorithms, providing radical timbre shifts that resemble the classic rhythmic effects of the Korg Wavestation. The synthesis capabilities of the DX200 far outstrip its utility as a groove box, so it's best to think of the onboard sequencer as frosting on an already rich cake.

Synthesizer/Sampler Hybrid (software)

NATIVE INSTRUMENTS
REAKTOR 3 (MAC/WIN; $499)

Native Instruments has established itself as one of the most exciting software companies around, and *Reaktor* 3 is one of the big reasons the company is on a roll. Among the many new features in this remarkable application are greatly improved performance; new sound-generating and processing modules; a more attractive, customizable interface; and better integration with the host program when running as a plug-in (VST or DXi).

Reaktor 3.0 provides dozens of synths and sample processors that can keep you busy for a long time. The examples are ready to play, with default samples assigned where needed, and many of the synths include internal sequencers that make them primed for the dance floor as soon as they load. When you're ready to create your own virtual instruments, you'll find new modules and macros that will jump-start your creations. The excellent

tutorials will help get you up and running quickly.

Check out the user-supplied patches at the Native Instruments Web page. You'll find everything from ARP Quadra emulations to wacky sample processors—all for free! You'll also find quality *Reaktor* Ensembles from third-party developers.

Whether you want to create your own virtual instruments or prefer ready-made synths, *Reaktor* 3.0 is a great resource. Processing samples, synthesizing sound, building tuning tables, and analyzing audio are all in a day's work for this sound-design powerhouse.

Synthesizer (keyboard, analog)

ALESIS
A6 ANDROMEDA ($3,499)

For lovers of analog synthesizers, 2001 was a great year, with more than a dozen companies churning out wonderful instruments. But one keyboard synth grabbed everyone's attention more than any other and deservedly so. The Alesis A6 Andromeda marries the classic sound of an analog synthesizer with state-of-the-art digital controls. The instrument is easy to set up and play, yet it provides a deep enough feature set to keep you programming for hours.

The Andromeda has 32 oscillators to help create a 16-note polyphonic, 16-voice multitimbral instrument. The oscillators, filters, and amplifiers are all under voltage control, and each voice has a dedicated output jack. Each voice passes through both a resonant, 24 dB–per-octave, 4-pole, Moog-style, lowpass filter and a resonant, 12 dB–per-octave, 2-pole, Oberheim SEM–style, multimode filter. Three rear-panel inputs allow you to run external signals through the filters. The A6 also includes an arpeggiator, a 16-step sequencer, and onboard effects, including analog distortion and digital chorus, delay, and reverb.

Real-time control is a hallmark of a useful performance synth, and in this respect, the A6 Andromeda delivers. The instrument includes Pitch-Bend and Modulation wheels and a ribbon strip, all of which are assignable to any modulation destination or MIDI continuous controller. The 72 knobs and 144 buttons give you further real-time control of your sounds. Add to this the three six-stage envelope generators, and you can see that the possibilities for modulation are extensive.

Manufacturing an analog synth of this caliber is difficult and expensive, but Alesis has risen to the challenge. Despite a difficult year for the company,

Alesis promises to meet the exceptional demand for this amazing keyboard synth. If you yearn for the sound and feel of a classic programmable analog synth, check out the A6 Andromeda while supplies last.

Synthesizer (keyboard, digital)

KORG
KARMA ($2,250)

Based on the synthesis engine of the best-selling Triton, the Korg Karma adds real-time pattern-generating and MIDI-processing muscle that dazzles nearly everyone who touches one. The Karma's complex dynamic algorithms respond to what you play, as you play, by altering tempo, timbre, chord structure, rhythmic density, and dozens of other parameters to produce new riffs, rhythms, and musical effects.

The result is an instrument combining the playability and timbral prowess of a keyboard workstation with the creative interactivity of a groove machine. The Karma is one of the first instruments that collaborates with players to create music in real time. Its collection of Generated Effects transforms performances with more than 400 variable parameters. Hands-on controls provide instant access to 16 simultaneous parameters that most synths don't offer. Korg's voicing professionals created hundreds of sounds that maximize Karma's response to what you play and how you play it.

Like the Triton, the Karma is fully expandable, with two slots for PCM-EXB expansion boards and another for an EXB-MOSS card. The Karma lacks the Triton's large touch-screen LCD and sampling capabilities, but with triple the Triton's processing power, the Karma responds to a slew of performance gestures. All of that technology enables players to execute more authentic arpeggios, realistic rolls, bodacious bass lines, and finer fingerpicking and strumming patterns than they can normally muster with just two hands.

With its evocative sound and groundbreaking playability, the Karma blazes a trail for future instruments.

Synthesizer (software)

NATIVE INSTRUMENTS
ABSYNTH 1.02 (MAC; $299)

Native Instruments *Absynth* is a semimodular software synth that extends the capabilities of the subtractive-synthesizer paradigm. *Absynth* is not an analog modeling synthesizer; the synth generates digital wavetables and supplies them to unique effects processors and an extremely powerful subtractive-synthesis engine—the concept at the heart of its powerful sound.

Absynth provides three dual oscillators per patch; each pair can act independently or interactively with ring, frequency, or amplitude modulation. A waveform editor lets you use the output of a modulated pair as the source for a single oscillator. Oscillators go beyond the typical sawtooth-, pulse-, and sine-wave choices: you can import the first few samples of an AIFF file, draw waveforms with the mouse, or adjust phase and amplitude settings for the 64 harmonics.

You also get several types of filtering options before the signal even reaches the two multimode filters. The package includes preset waveforms, and you can use any of them as an LFO source, so LFO modulation can be sine-wave smooth or jagged. Any wavetable can be used for the synth's waveshaping feature, which provides nonlinear distortion.

Absynth's 12, 68-stage envelope generators are a synthesist's dream. You can loop major portions of the envelope, and envelopes can retrigger in sync with tempo or number of beats. The synth gives you strong real-time MIDI control, too. For instance, you can control the amount of FM, filter frequency, and amplitude for each of the dual oscillators with your choice of Control Change messages, Velocity, and keyboard scaling.

Absynth made its debut as a standalone, single-MIDI-channel synth from Rhizomatic Software. Since its adoption by Native Instruments, the synth is multitimbral and applicable as a VST Instrument or MAS plug-in, as well. *Absynth* also supports ASIO, Open Music System (OMS), and FreeMIDI.

Synthesizer Workstation (software)

PROPELLERHEAD
REASON (MAC/WIN; $399.95)

A year ago, if you wanted a rack full of synths, sample players, drum machines, sequencers, mixers, effects processors, and patch bays, you could buy modules

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and connect them with dozens of cables, attempt to integrate several music programs and plug-ins, or combine the two approaches. Such solutions were often costly, and the software solution might require more processing power than your computer could handle.

Swedish software developer Propellerhead changed that with *Reason*, the world's first virtual music rack. Now anyone with a reasonably fast computer and a couple weeks' minimum wage can own a software-based stack of gear. Although each instrument is monotimbral, only your computer's horsepower limits the number of devices *Reason* can handle. If you run out of mixer channels, merely add another mixer. When the rack grows taller than your screen, just scroll down. If dozens of sliders, knobs, and switches grow unwieldy, reduce any module to the height of a half-rackspace with a click. Rearranging *Reason*'s virtual patch cords by pressing the Tab key to flip the rack

around and dragging the plugs where you want them.

This much virtual gear wouldn't be worthwhile if it didn't sound great: *Reason* sounds as impressive as it looks. It ships with over 500 MB of samples, drum kits, and loops, and you can download additional sounds and import sound files. Saving a song file preserves your studio setup, including all the sounds in the song. You can export songs as AIFF or WAV files.

Run *Reason* by itself or use ReWire to control it with a sequencer. You can play its modules from any MIDI instrument or control external instruments with its multitrack or monophonic sequencers. On playback *Reason* duplicates every control movement you record.

Reason runs on Mac OS or Windows and is compatible with a variety of audio hardware. It supports ASIO, MME, DirectX, and Apple Sound Manager and is capable of 24-bit, 96 kHz playback. ☻

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An article title enclosed in quotes indicates that the product was covered in a feature rather than in a review. All other entries indicate reviews of the award-winning version.

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Hit Men

Five heavyweight engineers reveal their chart-topping techniques.

By Michael Cooper

I t wasn't so long ago that different musical genres required, or at least lent themselves to, distinct production techniques and approaches. One didn't hear distortion guitar in '50s country music, for example, or open-tuned toms in '60s reggae. But in recent years, the lines have blurred. It's no longer surprising to hear scratching on a rock record or an inverse room reverb on a country-music production. Increasingly it seems that genres are defined more by instrumentation and musical content than by specific production techniques. In other words, anything goes.

I canvassed a broad cross section of mixdown engineers and producers—Elliot Scheiner, Tony Brown, Chuck Ainlay, Bob Clearmountain, and Al Eaton—to get their perspectives on producing hit records in this brave new world. Although some interviewees occasionally detailed production approaches specific to particular genres, more often the discussion turned to universal applications for getting a great mix. Here's what these studio veterans have to say about producing hit records.

ELLIOT SCHEINER

Elliot Scheiner has recorded, mixed, and produced many of the biggest names in pop, rock, R&B, blues, and adult contemporary music. From Toto, Fleetwood Mac, and the Eagles to B.B. King, Natalie Cole, and Dave Grusin, this multiple–Grammy Award winner has done it all. His recent projects include tracking for a new Steely Dan album and mixing sound for the IMAX movie *All Access: Front Row. Backstage. Live!* I caught up with Scheiner between mixdown sessions for Clint Black and Boz Scaggs.

Let's be up front. Scheiner approaches each song differently, but he admits that his mixes tend to be fairly heavy on drums. "When I mix, I set up the drum sound first," says Scheiner. "I start with whatever compression I'm going to add to that and then put in the bass. Usually, the bass and the bass drum, and maybe the snare, are the loudest things in the mix to start with. I try to make it so you can hear both the kick and bass very, very distinctly."

Scheiner frequently adds low bass-shelving EQ boost on the kick-drum track. "I'll do a fair amount of boost at, like, 30 cycles," he says. "I'll also throw a stereo compressor across the drums' stereo feed, and I'll use compression on individual drum tracks as well."

To avoid a muddy sound and to make room for electric bass in the mix, Scheiner usually applies some EQ cut to the kick-drum track at about 300 Hz. He tends to EQ bass tracks in the 200 to 250 Hz region, whether cutting or boosting, so that they're more clearly defined on small speakers.

Scheiner uses drum samples as seldom as possible, unless they're to control mic bleed. "I used some samples on the Eagles' *Hell Freezes Over* album [a live concert recording]," he explains. "Don Henley was singing, and a lot of times, I wanted to get some of the vocal leakage out of the drum tracks. I think I used two different samples, layering them with his original snare. But I never completely replace anything."

Double your pleasure. Scheiner says he often uses synthesizer tracks to perform unison doubles of acoustic guitar, string, and horn tracks to beef up pop and rock productions. Although double- and triple-tracking of electric guitars is less common now than in the past, it can provide a cool effect.

"I produced a Toto record a couple of years ago," Scheiner recalls. "Guitarist Steve Lukather's tendency is to double, sometimes to triple, a lot of his parts. He'll do a single eighth-note [ostinato] part and then double it and then VSO [varispeed the doubled part] up. Then he will triple it and VSO that third track down." Scheiner says the individual parts may be panned apart or to the same location in the stereo sound field.

According to Scheiner, most contemporary rock-lead-vocal tracks employ few if any time-based effects (reverb and delay, for example). However, they are often recorded with two preamps chained together in series to create a little distortion. "I've come across a lot of that lately," he says. "I did it 30 years ago, not only on vocals but primarily on guitars."

Scheiner doesn't like to stack tons of background vocals in his productions. "I can't imagine anything that needs 32 channels of vocals," he says.

The days when an entire band played together to create a live take are a distant memory for most engineers. First-pass, basic tracks for a typical pop or rock session now consist of bass, drums, one guitar, a reference vocal (that later will be replaced), and sometimes keyboards. Nevertheless, on Steely Dan's newest record, the band laid down tracks the old-fashioned way—all at once. To keep down bleed in the small room the group recorded in, Scheiner kept the bass-guitar amp set to a fairly low volume and

Elliot Scheiner's engineering genius on several Steely Dan albums won him multiple Grammy Awards.

housed the electric-guitar amp inside stacked gobos.

Squeeze me. Scheiner almost always straps a stereo compressor across the stereo bus when mixing. "I love the sound of the Neve 33609," he says. "I don't use it for compression as much as for the sound of the compressor." He uses a maximum of 2 to 3 dB of compression on peaks, with a 1.5:1 ratio.

Scheiner generally avoids mixing on large monitors, relying instead on industry-standard Yamaha NS-10Ms to get the perfect stereo mix. "I can't live without the NS-10Ms," he says emphatically. "They're generally all I use to mix in stereo. I monitor very softly, at maybe 65 dB SPL." If he's adding effects to individual tracks, he monitors louder. But once he gets everything in, he listens softly.

TONY BROWN

Tony Brown is not just the president of MCA Records in Nashville; he's also one of country music's hottest record producers. Brown's discography reads like a Who's Who of country music and includes albums by Vince Gill, Mark Chesnutt, Wynonna, Reba, George Strait, Trisha Yearwood, and Alabama. The prolific studio veteran has produced or coproduced in excess of 100 albums, more than 40 of which went

Hit Men

Gold, Platinum, or multi-Platinum, and which together spawned more than 90 No. 1 singles. The recipient of multiple Grammy, Country Music Association, TNN, and Billboard Awards talks about how country music has evolved during the past several years.

Back to the future. "In country music, '80s rock is alive and well," Brown says. "Some of the guitar solos and other sounds are very reminiscent of the late '70s and early '80s.

"Drum and percussive loops are especially big now," he adds. "I think I'm one of the last holdouts, but I finally succumbed to using them. But some people are overdoing it. In country, when they use some of these loops that you'd hear on a hip-hop or pop record, it just sounds really cheesy to me. They're trying to be hip, but in fact, a lot of times they're being unhip. I've always thought that the song has to lend itself to that kind of treatment, and the artist has got to be able to pull it off. If the artist can't pull it off and the song is really a country song dressed in pop trappings, it just sounds like a record with no home."

Tony Brown, president of MCA Nashville, is one of the most sought-after producers in country music.
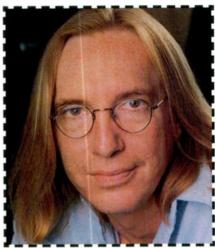
Brown has a reputation for working fast in the studio. His approach on the past few George Strait albums is typical of the way he likes to work. The first recording pass usually includes drums, electric lead guitar, acoustic guitar, steel guitar, fiddle, keyboard, and upright or electric bass. "We cut the tracks live in just four days," he says. "The only things we overdub are background vocals and strings. I don't go in and start moving parts around. We fix anything that needs fixing while [Strait] is there. It's a performance kind of record.

"I'm a big rhythm section kind of guy," he continues. "Most of the acts I worked with when I first started producing—including Jimmy Buffet, Vince Gill, George Strait, Wynonna, and Reba—like recording with the full band. So when you go back into the control room and hear the tracks, you're almost hearing the finished record."

The vocal rules. If country music has a prime directive, it's "don't step on the vocal." That means, among other things, keeping the arrangement tidy and the effects levels under control. "Country music favors singers with chops, those who can do tricks with their voices," says Brown. "And the lyrics telling the story are so important—they have got to cut through. If you put too much stuff around the voice, it just soaks it all up. I'm into the clean approach." Skilled musicianship is also a must. "To me," Brown says, "country music has always been about musicianship and a voice."

Even though the drums and guitars on contemporary country records are typically louder than those on past recordings, the vocal must be out front enough for the lyrics to be heard clearly; the drums continue to play a supporting role. "They typically hit on the two and four," Brown says, though he notes that statement can be an oversimplification. Any drum fills, he adds, are usually found going into the chorus or leading into the bridges.

Brown says that you can have a country record without fiddle or steel guitar, but acoustic six-string guitar is less dispensable. "Ninety-eight percent of the time, a country record will have an acoustic guitar," he says. "The acoustic is still the

Chuck Ainlay has worked behind the console for artists as diverse as Vince Gill, Melissa Etheridge, and Peter Frampton.

best thing to glue and hold the track together with the drums. It can be either strummed or used to play arpeggios. In either case, the acoustic player on a country session has to be really, really good." That's because the same guitarist is often called upon to use a wide variety of techniques, from bluegrass licks to open tunings and so on.

Keyboards are frequently used as "sweetener" tracks in country productions. Those tracks are usually mixed in at a low level for a subtle effect. "I'll add an organ or a really subtle synth pad that you can feel more than you can really hear in order to add some depth to the track," Brown explains. "That can give a country track a sheen and make it sound less brittle."

CHUCK AINLAY

Although he started out recording local rock acts in the Nashville area, Chuck Ainlay has become one of country music's hottest engineers. His song-by-song approach to mixing—as opposed to following an established country-music production route—kicked his career into high gear because Nashville was looking to hear something new. A Grammy and TEC Award nominee, Ainlay has worked behind the mixing board for a diverse range of artists including Vince Gill, John Anderson, Wynonna, Trisha Yearwood, Steve Earle, Melissa Etheridge, Peter Frampton,

Hit Men

Dire Straits, and Mark Knopfler (whom he also has produced).

Whoa, trigger! "I did the first recording in town that simultaneously triggered drum samples with live drums," Ainlay recalls. "But I haven't pulled out my sample rack in the last four years to do any of that. You've got these great drummers playing, and you lose the inflections of what they're doing by putting in samples. It's still done sometimes, but there are techniques we've all learned to make live drums have the same kind of impact."

One such technique is to engage two or more ratio buttons simultaneously on a Urei 1176LN FET compressor to create a pronounced point on the attack of a kick- or snare-drum track. The output of the compressor is then gated and mixed in with the original dry track at the mixer. "If I am going to do a squeeze like that," Ainlay says, "I definitely would mult the compressed track, bring it up on another channel, and gate it so I don't have all the background noise."

Bob Clearmountain's masterful rock mixes have earned him legendary status among industry pundits and colleagues.

If any generalization can be made about country as compared with rock productions, it's in how the bottom end is mixed. "With rock records," Ainlay says, "that whole bottom area can be muddy with mic bleed and ambience from the room and actually contribute to the sonics. Country music, though, has always been well defined on the bottom end. We go for a really percussive, full, and punchy low end. In the past five years, starting with Shania Twain and Mutt Lange, her producer, it has gone away from that a bit. But I think it will be back."

Regarding kick-drum sounds on country records, Ainlay says, "We probably put a bit more snap in the bass drum, really [cutting] the EQ at 400 Hz and putting more 4 kHz stuff in there."

Sibyl in the studio. If one instrument tends toward multiple personalities in country-music sessions, it's the acoustic guitar. "Open tunings might be played along with the regular tuning," Ainlay notes. "There are also capo'd tunings, or you can play up in a higher register on the acoustic. There's also a high-strung acoustic, which rather than using round-wounds uses single strings from the fourth string on up. Different tunings can also be used with the high-strung that puts that 12-string sound in there."

Ainlay explains that the high-strung guitar is usually overdubbed on top of the original acoustic-guitar track. "On a double pass, you can slightly varispeed the guitar so you get a wider-sounding thing.

"I don't think there are any production techniques done in rock that aren't done in country," Ainlay says. "They're just maybe not as blatantly obvious in country. Synthesizers are used often, but probably with more organic rather than synthetic sounds. Keyboard sounds are in about everything you hear in country music these days. MIDI pads with acoustic piano are used a lot. Hammond B-3, the real thing, gets used on a lot of songs. We also see a lot of Wurlitzer still being used, but maybe not as much Rhodes. But Moog-style bass sounds and 808 drums—all that stuff, layered with real sounds—gets into country records these days."

But just because synthesizers have invaded country music doesn't mean the tracks are sequenced. "It's usually a more performance-oriented rather than sequencer-driven track," Ainlay says. "However, most tracks are played to a click."

Ainlay says that the majority of country sessions start by simultaneously recording "a pretty full rhythm section consisting of bass, drums, a couple of guitars, fiddle, steel, and at least one keyboard, if not two. Generally, most songs are cut within three or four takes."

Although some country artists take their own bands into the studio, backing bands of top-notch session players are more commonly used. Background vocals, strings, and pads are regularly overdubbed.

Living in harmony. When a production calls for maintaining the singer's personality and vocal nuance, as is typically the case in country music, Ainlay tries not to use too many effects on the lead vocal. "There's usually a little plus/minus harmonization stuck on the lead vocal to spread it out and make it sound warmer," he explains. "But mostly that's a subtle thing to try and widen the vocal."

Ainlay always uses a stereo compressor across the mixer's stereo bus. "If I leave it to somebody else to deal with," he says, "how am I going to know that what I did is going to work when it gets compressed? I'd like to have a hand in that. Usually, I start with the compressor out [bypassed] and try to get balances that work without a mix-bus compressor. Then I'll switch the compressor on and rework things to get that ultimate groove before I turn on the automation."

BOB CLEARMOUNTAIN

Since 1972 master engineer Bob Clearmountain has worked his magic on more than 200 albums and singles. The three-time Grammy Award nominee and ten-time TEC Award winner is best known for his winning mixes for pop and rock luminaries such as Bryan Adams, Bruce Springsteen, the Rolling Stones, INXS, Crowded House, David Bowie, and Shawn Colvin.

Center of attention. "With pop records in particular, everything revolves around the voice," Clearmountain says. "It's by far the most important aspect, and it sets the tone for the mix."

Clearmountain prefers to mix on Yamaha NS-10Ms (which have an inherently high bass cutoff), and he makes no bones about his preference for mixing on small monitors. "I'm better at judging bass, bass drum, and vocal levels on my little Apple-powered computer speakers," he says. "I don't own any big monitors. The biggest ones I've got are the KRK Exposé 7s. I've actually had more problems trying to judge bottom end on big soffit-mounted control-room speakers."

In discussing how drum sounds change every few years, Clearmountain says: "People used to go for the big snare-drum sound, but that's really out of favor nowadays. Now it's more of a small papery sound that people like. That's a big thing for pop records. But I tend to think that that stuff is not so important to the mix. What's critical is the vocal and getting the bottom end in the right perspective."

Clearly the bottom. "I don't have a huge preference," Clearmountain says, "but I like to do everything—record and mix—digitally; especially now, with the new Apogee gear, digital sounds amazing. To me, analog is unpredictable; it does that funny thing to the bottom end. You work really hard on the bottom to get it exactly right, and then you play it back on your analog tape, and it's like, 'Oh, what happened there?' The storage medium is making decisions about what the bottom end should sound like.

"If I'm having trouble with the mix because it's sounding cloudy and muddy on the bottom end," says Clearmountain, "I'll turn the bass drum and bass off to determine what else might be booming away. You might not realize that there are some frequencies in there that aren't really musically necessary to your mix. With piano or guitar, you're mainly looking for the mids and top end to cut through—the low-end stuff is just getting in the way. So you clear some of that out with some filtering, and then

when you solo it, you think, 'Man, now my piano or acoustic-guitar sound is too thin.' But when you let go of your solo button, all of a sudden the bass sounds so much better, and you're not really missing that stuff from the other instruments. Now the mix sounds louder, clearer, and fuller. Of course, you can't EQ the other instruments too drastically, or you'll lose the warmth of the mix."

Clearmountain points out that what works in one section of a mix might not work elsewhere: "On a mix where there is just a piano and vocal intro, for instance, I'll bring the piano up on two sets of faders. One fader will be for the intro, for which you have the piano sounding nice and full. But once the drums and bass and the rest of the band kick in, I'll go to the fader with the filtered piano sound. You don't notice when that change occurs, because all of the other instruments are taking up that space in the bottom end."

Question of balance. "In order to determine if a mix is working or not, I always try to picture it visually," says Clearmountain. "All of the instruments and everything else in the mix have to relate to each other on some level. Either there's a huge contrast on a track to make it sound more striking, or the tracks sound like they're in a comparable space but with a slightly different perspective. But to go through and say, 'This sounds kinda nice on the voice, and this other thing sounds nice on the background vocals, and this other thing sounds nice on the snare drum'—with that approach you just end up with all these separate things going on, and they don't necessarily relate to each other."

Clearmountain also spoke about how beginning engineers often make the mistake of adding reverb to everything in an attempt to make the mix sound big. "You get size from contrast," he says. "To make something sound big, there's got to be something in there that sounds small."

To further beef up his mixes, Clearmountain regularly places a stereo compressor across the mix bus. He always listens postcompressor and post-A/D converter throughout the mixdown process to take into account the contri-

Triple-threat Al Eaton also appears as composer and engineer on many of the hip-hop records he produces.

butions of that gear to the overall sound.

Pet peeves. Clearmountain is quick to hit the mute switch on any track that doesn't contribute meaningfully to a mix—unless, of course, the producer or artist insists that it be left in. "Wind chimes get the cut switch," he says. "There's no room in records for that. It's just noise. Scratching should get the same treatment, as far as I'm concerned. It just sounds silly on rock records, like they're trying to be hip and cool and sound just like a rap record. It's as if a decree came down from the label: 'We should sell some more records. Let's put a scratching noise on there.' But that's not rock 'n' roll.

"I've gotten to like loops, which I hated for years and years," says Clearmountain. "They'll probably go out of style soon, now that I like them. Like anything else, they're very useful as long as they don't distract from the song."

AL EATON
Known by many as the godfather of hip-hop, Al Eaton played on the first Northern California hip-hop record in 1981. Highly versatile, Eaton often writes, arranges, records, mixes, and produces records from start to finish. His credits include Gold and Platinum hits with hip-hop artists Master P, Too Short, Silkk the Shocker, Mia X, and Mr. Serv-On as well as numerous productions and remixes for Arrested Development,

Hit Men

Queen Latifah, Ice-T, Zhané, and others.

East meets west. Our conversation got under way with a discussion of hip-hop's jazz-tinged East Coast roots as differentiated from rap's West Coast origins and funk influences. Eaton says that the distinctions between rap and hip-hop have mostly faded, making labels more confusing than helpful. Yet subtle differences between East and West Coast styles persist.

"East Coast rappers are usually more into rhyming skills and wordplay, whereas West Coast rappers are more into the story," Eaton says. One thing common to rap and hip-hop is an up-front mix of vocals and drums. "Kick and snare are always slapping you in the face, and everything is usually really, really dry," says Eaton. "On a lot of hip-hop and rap records, the vocal has absolutely no effects. It'll be right up in your face."

One reason for that is that many records are geared toward being played in big clubs. "Usually, the bigger clubs already have a lot of ambience," Eaton explains, "so you really don't need to do anything to the track. The drier the record is, the more impact it has when played in the club."

Breaking new ground. Although the ubiquitous Roland TR-808 kick is a standby in urban music, Eaton has tried to steer clear of using it in his most recent productions. Instead, he often experiments with substituting synth sounds. "In some cases, I'll use a string patch played on the low end," he says, "and then I'll do some tonal stuff to it so you don't recognize it as a string line." He may or may not change the envelope of the string sound, depending on the song.

Eaton often favors dropping audio samples into Mark of the Unicorn's *Digital Performer* over using a performance approach for laying down drum tracks. "I'll import drum sounds and place them within the quantization grid so that they're playing the drum pattern I want," he explains. Eaton accomplishes that by first importing the audio samples into the Soundbites window and then dragging the sound bites (consisting of single sounds) into the Graphic Editor window. "You can use the Groove Quantize feature to give it swing or whatever," he says.

The drums and bass have a well-defined relationship in urban-music productions. "In a lot of rap songs now, the rhythm of the kick drum and bass line are pretty much the same, and they kind of follow and meld into each other to sound like one instrument," Eaton says. "Nine times out of ten, the snare is going to be on beats two and four."

"Most country songs cut within three or four takes."
—Chuck Ainlay

Sweetener-track instruments commonly used in modern rap and hip-hop productions include cabasa, shaker, strings, and the "worm," a high sine-wave-like sound, typically from a mono synth, playing a portamento-inflected melody line.

Eaton doesn't hesitate to use stereo compression to give a mix an aggressive edge. "The big thing now in urban records is loudness," he says. "A lot of it is not about tone per se. It's about having the loudest record you can possibly have. I don't want to fall into that, but if you want to compete, then you kinda gotta go there."

Michael Cooper *is the owner of Michael Cooper Recording, located outside the beautiful resort town of Sisters at the base of the Oregon Cascades.*

Taming of the Shrill

Notes on recording harmonica from a bevy of heavy breathers.

By Brian Knave

Although the harmonica can be played sweetly, it's not intrinsically a sweet-sounding instrument. Rather, it ranks high on the list of potentially annoying musical instruments—right up there with the violin. I learned that as a child when I became serious about practicing the harmonica. If I took out my harp, my friends scattered. I might as well have been carrying a weapon.

Then again, as those who frequent open-mic and amateur nights know, an amplified harmonica in the wrong hands practically is a weapon—it can clear a room in a hurry. No, this instrument's sweetness must be coaxed out. Perhaps that is not so surprising, considering that the harmonica is constructed of tiny metal reeds sandwiched between copper plates and tin covers. Throw in a gutful of wind, and you have a surefire recipe for a bright, piercing tone.

Not unexpectedly, some mics don't take well to the harmonica's shriller timbres. That's one reason ceramic- and crystal-element mics such as the Shure "Bullet" and the Astatic JT 30 are popular among blues-harp players: their severely limited frequency response, especially on the high end, really warms up the sound. Even a beginning player can readily produce a round, fat (albeit distorted) tone when armed with a Bullet mic patched through a tube amp.

But this column is not a tutorial about recording blues harp. The task there, after all, is essentially the same as for recording an electric guitar: you mic the amp rather than the player. (For really cool tips on recording amps, see "Recording Musician: Recording Electric Guitar" in the October 1999 issue.) The focus here is on recording acoustic or unamplified harp, specifically diatonic

A vintage AKG C 12 tube mic captures the intricate stylings of bluegrass harmonica wizard Mike Stevens.

and chromatic models. That may sound like a breeze to the uninitiated, but it's not always so straightforward. I've been playing diatonic harmonica for more than 30 years and have racked up countless hours recording the instrument, both live and in the studio. Of the many instruments I've learned to record, the harmonica has proved to be one of the more difficult to capture satisfactorily.

Besides tackling the main task of taming the instrument's naturally penetrating tone, the recordist should also be prepared to work with the instrument's broad range of tones and dynamics. Accomplished players will use not only their tongues, throats, jaws, and breaths to shape and color the sound but also their hands, sometimes fully enclosing the instrument. Vibrato and tremolo are both common, as are note bends, wah effects (similar to those produced by a muted trumpet), and other manipulations. The engineer may also have to deal with various incidental sounds—

grunts, gasping breaths, air leakage from the nose, and so on. But perhaps the biggest challenge is the not-so-accomplished player, the aforementioned weapon wielder, particularly if he or she hasn't learned to temper the harmonica's native bite to produce a smoother, more musical tone.

There's no right way to record harmonica; different players have different sounds and approaches, and what works for one may not work for the next. So to broaden the scope of this piece beyond my own experience, I interviewed four of the finest and most recorded harmonica players on the scene today: jazz master Jean "Toots" Thielemans (www .tootsthielemans.com), pop giant and

On his latest release, Norton Buffalo opted to record all the harmonica parts unplugged (without amplification) to "bring out the true beauty and power of the harp." A Neumann U 87 is his studio mic of choice.

harmonica manufacturer Lee Oskar (www.leeoskar.com), multigenre maestro

Norton Buffalo (www.norton-buffalo .com), and bluegrass sensation Mike Stevens (www.mikestevensmusic.com). Check out their Web sites, especially if you aren't familiar with their work. They are consummate musicians and masters of the harmonica, and their respective recording and performing credits, awards, and other distinctions are enough to make your head swim.

THE BALL ROLLING

Faithful capture is the general idea when recording any acoustic instrument. First, have the musician play the part while you listen carefully to the sound of the harmonica in the room. That will give you a sense of the player's abilities as well as the overall tone he or she produces. The ten-hole diatonic harmonica comes in several brands and models. Different brands can sound quite different from one another, and materials and quality of construction can also affect the tone. A harmonica with a

wooden comb, for example, may sound warmer than one with a plastic comb, and a cheap harmonica may sound thinner (and leak more air) than a pricier model.

Another thing to determine up front is whether the player will record *open mic* (that is, next to the microphone with the mic mounted on a stand) or *handheld* (while holding the mic). Generally, the former provides a more natural tone; the latter, because it requires cupping the mic and harmonica together, largely closes off the sound from room ambience and, depending on the mic, increases bass response due to the proximity effect (both potentially helpful things, by the way).

FIG. 1: One helpful trick is to set up two different mics side by side and record the harmonica simultaneously to two tracks. That not only fattens the sound but also allows for "organic" equalization—the two tracks can simply be blended to taste during mixdown—alleviating any need for EQ. Shown are two of the author's favorite mics for recording harmonica: the Microtech Gefell MT 711S condenser and Sennheiser 441 dynamic.

Thielemans always holds his mic, so the room has little or no effect on his

recorded sound. If you are recording open mic, however, the acoustics of the space can have a considerable effect on the sound of the tracks you record. Although nearly anything can be made to work—my studio is quite small and acoustically dead, yet the harmonica tracks I've recorded in it sound just fine—I usually prefer a medium to large wood-paneled room with a hard-wood or tile floor and a spacious but not cavernous sound. In other words, I

want a room that adds favorably to the harmonica's tone, thus inspiring me to play. Stevens concurs: "I like a room with a high ceiling and a bit of reflection—you don't want it too dry. Basically, the room should let the harmonica sound natural while adding some ambience to the sound."

The distance from the instrument to the mic also figures into the equation: the farther back it is, the more the acoustics of the space affect the sound.

Lee Oskar, who played with the group War for 23 years, has one of the sweetest harmonica tones you could ever hope to record. He and other former War members (sans Lonnie Jordan) have recently reformed as SOB—short for Same Old Band.

High-frequency capture tends to diminish with increased distance from the mic, and the tone becomes more diffuse (something classical-music players often appreciate). That is useful to know if you're having a problem with the harmonica sounding overly bright or strident and you don't have another mic (or player) at your disposal. Foam-rubber windscreens and pop filters can also help reduce troublesome highs in such cases. Once, to quell the edge of a particularly bright condenser mic, I fitted the mic with a windscreen and two pop filters. That noticeably altered the high-end response, and though I still wasn't in love with the sound, I was at least able to get some usable tracks.

IT'S THE MIC, STUPID
The better the player and the smoother and warmer his or her tone, the more leeway you will have with microphone selection. Nevertheless, few mics really excel at recording harmonica. In general, seek those that downplay the instrument's reedy highs and bolster its warmer timbres. "Harmonica can be very bright and thin sounding if you use the wrong mic," says Oskar. "What I listen for is a nice, open, really warm yet
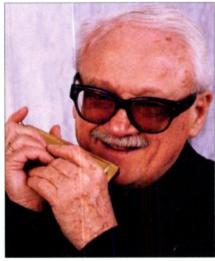
transparent sound. The mic has to sound natural and be capable of capturing all the tone, the breath, everything, but without sounding harsh."

"I can walk up to a mic and pretty much instantly tell whether it's going to be good or bad for harmonica just by talking into it," says Stevens. "As soon as I hit one that emphasizes the chest frequencies—those really low animal sounds—that's the mic I want. What I like in a mic is lots of fat low end, the mids [1 to 5 kHz] kind of scooped out, and a very extended but smooth top end, just to give it some air. Even a bad mic equalized like that can sound pretty decent."

The four players I talked with expressed different tastes in mics. Buffalo has used a Neumann U 87 to record most of his acoustic harmonica work to date. "I've also used U 67s, and generally, I like them better, but 87s are a lot more available," he adds. "The 67

DIRK HERMANS

Toots Thielemans, the legendary jazz musician whose playing has almost single-handedly defined the genre for chromatic harmonica, records all of his tracks with—surprise—a handheld Shure SM58. "It has been a good, faithful friend," he says.

is a tube mic, so it sounds a little warmer. But that added warmth can also bring out more of the breathiness, which can be a problem in some cases."

Although also a longtime fan of Neumann mics, Oskar's current favorite for open-mic recording is the BLUE Bottle (a high-end hand-built tube mic) fitted with the BLUE B7 (classic cardioid) capsule. "It's an absolutely incredible mic," he says. "[It's] extremely clear sounding and right in your face." For handheld recording (as well as for playing live), Oskar uses the beyerdynamic M 160, a ribbon mic. "The ribbon component is probably the best thing for recording harmonica. It's similar to a reed structurally, the way it moves, and it's very warm sounding."

Of course, high-end mics aren't always the ticket. Thielemans, no stranger to the best studios in the world, has many times encountered engineers determined to use their finest, most expensive microphones to capture his beautiful tone on the chromatic harmonica. His favorite mic, though, is the Shure SM58. "What's important is not hi-fi but my-fi," says Thielemans. "I've tried everything, and the SM58 is what works best for me."

Thielemans has no problem with the engineer equalizing the harmonica track

so it better fits the mix. His manager, Dirk Godts, has picked up the basic EQ curves that are sometimes required to get Thielemans's sound in live situations. "I almost always reduce the really high frequencies a bit, because the instrument is already so sharp sounding that it doesn't need them," Godts says. "Then I add some mids to give the instrument a bit of belly, and I also cut the very lowest frequencies."

DOWN TO THE WIRE

Of the players I talked with, Stevens was clearly the gear nut—he even specified his favorite mic preamps (the Telefunken V76 and "Great River preamps with the output transformers"). For condensers, he suggests "anything with the Neumann M7 capsule." That includes, naturally, certain Neumann mics such as the tube models U 47 and M 147. But it also includes some models from Microtech Gefell, among which Stevens recommends the MT 711S. I reviewed that mic several years ago and proclaimed it "the best-sounding solid-state condenser I've ever used on harmonica" (see "Attack of the Cardioids" in the September 1998 issue).

For dynamics, Stevens favors the Sennheiser 441 (one of my favorites), the Shure Beta 58, and the Audix OM-7. "The Beta 58 is an old standby that always seems to work well," he says. "[The OM-7 is] great for handheld recording when you need something really clean. It has almost no proximity effect, so you can cup it and get no bass boosting." Stevens also suggests the Electro-Voice PL6, a discontinued cardioid, which he also uses handheld.

Like Oskar, Stevens is enthusiastic about ribbon mics. He particularly likes vintage models such as the Shure 330 "Unitron" and the RCA 44BX, "as long as it's in good shape," he says. Additional ribbon picks are the Coles 4038 and the Royer R-121. "The Royer is one of my favorite mics at the moment," Stevens says. "It can be a bit dark, but it sounds really great, especially if you have a good-sounding room to record in: the figure-8 pattern causes it to pick up a lot of room."

After talking with Stevens, I spent a day in the studio doing some comparison testing. I own a Royer R-121 but had never used it for harmonica. Curious about some of Stevens's other comments, I set up the R-121, the Microtech Gefell 711, a Sennheiser 441, a Shure SM58 and a Shure Beta 58. I recorded several harmonica tracks and examined the results. The R-121 tracks were indeed dark but natural sounding, without a hint of high-end grit. Of the three other dynamics, the 441 is my favorite. It is unrivaled for capturing punchy, hornlike sounds from the diatonic harp. The tone is round, smooth, and solid, with lots of meat on low *chucka-chucka* rhythm parts and fully present yet nongrating highs.

For my tastes, the SM58 and Beta 58 work better handheld than open, predominantly because of the proximity effect, which helps fatten the tone. Either way, though, they sound a tad covered and even somewhat compressed as compared with the (much more expensive) Sennheiser 441. Of the two Shure mics, I prefer the Beta 58 on harp because of its smoother highs and overall warmer sound.

Although it's not something the engineer ordinarily has any say about, it's worth noting that the pitch range or key of the harmonica can also play a role in mic selection. In general, the higher the pitch of the harp (or the section of the harp being played), the more likely I am to use a dynamic mic, particularly a ribbon. On the other hand, low-tuned harmonicas or low-chord rhythm parts may benefit from the additional high-end response a condenser mic can provide.

POSITION ON THE MATTER

For close, open-mic-style recording, mic placement is critical. The trick is to minimize extraneous noise by positioning the mic directly in line with the sound coming from the instrument. Watch and listen to the player to make that determination.

When Buffalo recorded his first solo album, *Lovin' in the Valley of the Moon* (Capitol Records, 1977), the engineer hung the mic above him in the manner frequently used for recording singers. "The breathing and grunting noises

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coming out of my nose were almost louder than the harmonica," he recalls. "Being a bit green about recording at the time, I didn't know how to deal with that. We ended up putting masking tape across my nose to keep the breathing out!" That experience led Buffalo to question where the instrument sound was primarily issuing from. "After realizing it came out more from the bottom of my hands, I lowered the mic so the diaphragm was aiming up from beneath my hands. That position gets rid of most of the nose sound, and it allows me to work the mic very close."

Buffalo relates an interesting tip about using pop filters: "I find that U 87s tend to close down with too much humidity, so during the winter or anytime it's moist or rainy out (or humid like in the South), I'll put up a pop filter to help spare the mic. But when it's dry, you can play close on an 87 with no problem—and the way I play, the closer I get, the more beautiful and warm the tone is."

DOUBLE WHAMMIES
A useful technique I often employ is double miking. I position two different-sounding microphones, typically one dynamic and one condenser, side by side as close together as possible, with the diaphragms in the same plane (see **Fig. 1**). Each signal is routed to its own track, and the two levels are closely matched. The player directs his or her performance to the center of the two mics from about six inches back so that the mics hear the instrument equally. The result is two distinct tracks that can be blended to taste during mixdown, typically in lieu of any EQ.

Oskar offers a different way to lay down two simultaneous tracks. "Sometimes I'll hold the beyerdynamic M 160 in my hands and play into the BLUE Bottle mic at the same time," he says. "That way I get the body from the hand-held mic and some presence and room sound from the condenser. Later I can blend the two tracks however I want."

Stevens offers another, more involved double-whammy technique: "I split the signal coming from the mic and send one half to one track and the other to a really fast, clean power amp and a great
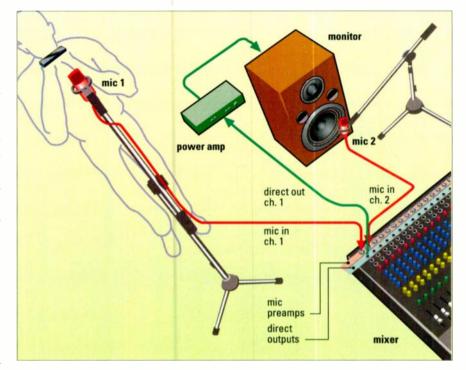
FIG. 2: You can route signals several ways for Mike Stevens's double-whammy approach to recording harmonica. Here, the initial mic signal (mic 1) is split at the console and sent through a direct output to a separate power amp. The amp powers a monitor speaker which is miked (mic 2) and routed to a second console channel, allowing for capture of two simultaneous but different-sounding tracks.

speaker—a reference monitor, for example. The monitor is turned up in the room so I can hear it and miked from about 10 to 20 inches back. The second mic signal is sent to another track, which gives you two tracks to mix together. It sounds wonderful; you get a totally acoustic sound that's as big as a house" (see **Fig. 2**).

Stevens discovered another cool trick by accident on a session for which he had to switch back and forth between amplified harp (miked with a Shure Bullet) and acoustic harp. "When it came time for the acoustic stuff, I turned down the volume pot on the Bullet, but I didn't have time to set it down. As I cupped the Bullet and played acoustically into the other mic, a U 47, I got this unbelievably sweet sound. Now, on certain sessions, I'll actually put the Bullet in my hands like that—unplugged—just to shape the tone as I record acoustically."

AN OPEN EAR
Stevens volunteers the following advice about the best type of headphones to

use when recording harmonica: "I recommend open-ear rather than isolation-type phones. With isolation phones, you can sometimes hear the sound of the harp radiating up through your jaw, which freaks out a lot of harp players and tends to make them play flat. Open-ear-type phones let you hear yourself in the room, which makes it easier to stay in tune."

If you don't have access to open-ear phones, Oskar recommends the old one headphone on, one headphone off trick. "When using great mics, the sound can sometimes be so overwhelming that it's easy to overplay," he says. "In that case, it really helps to be able to hear yourself rather than just what's coming through the mic."

You Ought to Be in Pictures

Get the big picture on marrying your music to movies and television.

By Michael A. Aczon

From its humble beginnings with pianists accompanying silent movies to the platinum-selling soundtracks for today's films and television shows, the union of sight and sound has developed into a multimillion-dollar business.

Film soundtracks frequently top the music charts, music videos promote movies and the songs in them, and popular songs pop up constantly in TV shows and commercials. Sometimes previously unknown artists gain a national audience by contributing to a movie or TV production. For example, Elliott Smith had a small cult following when his song "Miss Misery" appeared in the movie *Good Will Hunting;* the song garnered Smith a hit single and an Academy Award nomination. Vonda Shepard vaulted to popularity thanks to her appearances on *Ally McBeal* and her prominence on the show's album soundtracks.

The century-long relationship between music and moving pictures shows no sign of abating anytime soon; in fact, it is continuing to grow at a steady clip. The question is, how can you successfully take your music from the studio to the screen?

BUSINESS BACKGROUND

As a musician, songwriter, or producer, you can pursue a variety of avenues to break into this area of the entertainment industry. Whether it's pushing your catalog of independently released records or songs, writing songs for a specific project, or producing a compilation of your best background music, some basic information can go a long way toward getting you a break into the business.

First of all, the business aspect of

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transferring your music to the screen is pretty straightforward. Movie and TV production companies pay negotiated fees to use musical compositions and master recordings. The fees vary widely based on factors such as the type and length of use, the importance of a song to a production, and the size of budget set aside for music. Performance royalties are generated every time a song from a movie or television program is broadcast on the radio, television, or the Internet. Another major component of the music-and-visual mix is the role of the record industry. Simultaneous cross promotion, such as the music video tie-in, by record companies and motion picture distributors often results in a tremendous sales boost for soundtracks.

SUPPORTING CAST

Knowing the various players involved in getting music into a movie or TV project is key in your efforts. Whether a production is big or small, the roles of key personnel remain the same.

Music supervisor. This is the person in charge of finding the right music for a production. Sometimes that is simply a matter of finding preexisting commercial recordings of songs that fit. Other times music supervisors have to find composers to write specific musical pieces or producers to rerecord existing songs to better fit the visual. After finding the right music, the music supervisor then negotiates the various rights to use the masters and compositions with the owners of those rights.

Music publishers. Publishers are the frontline sales force that gets songs to music supervisors. Some music publishers specialize in film and television placements; others, particularly the larger publishers, have separate film and television departments. Music publishers constantly read trade magazines, meet regularly with music supervisors and movie directors, and make it their business to know what movies and television programs are in production and what stage of production they have reached. It is the publisher's job to provide the right song or composer for a production at a moment's notice.

Artist and repertoire (A&R). The A&R personnel at record labels are also involved in looking for projects that might be good links to record sales. Labels pay a fee to the movie or television production company for the soundtrack rights to a project, knowing that sharing an audience with the movie or television show can lead to a hit record.

Music-library companies. These companies provide, to music supervisors, prerecorded music ready to use and license once the right fit is established. Music libraries typically own the masters and compositions included in their CDs, which is attractive for music supervisors because negotiating for both uses is one-stop shopping.

Online music licensing. These services are a relatively new addition to the field. A high-tech hybrid of the traditional music-publishing business and music libraries, the business model for online licensing services is being established. Some companies own all the rights to the music and masters they represent; others represent owners of compositions and masters on a nonexclusive basis. When music supervisors find songs or masters online that they want to use, the licensing service negotiates the fee for such use and then pays a prenegotiated portion of the fee to the owners of the masters and songs.

Performing-rights organizations. These organizations license and collect the fees for the songs' performance rights. The U.S. performing-rights organizations—ASCAP, BMI, and SESAC—have entire departments of specialists who negotiate licenses with the users of songs in their catalogs for television, film, and new media.

FOCUS IN

Narrowing your view of what you want to do in the film and television business will determine your approach on getting in. If you are a recording artist, your focus is probably more on exposing your records to a larger audience than on creating music specifically for a film or television program. Because labels are in the business of getting their artists heard, soundtracks are a

Elliott Smith was a critically acclaimed singer and songwriter with a small cult following when he contributed six songs to the *Good Will Hunting* soundtrack. One of the tracks, "Miss Misery," earned him an Academy Award nomination.

great opportunity to establish up-and-coming artists. If you are fortunate enough to be signed to a label, talk to your A&R and promotion departments about seeking those opportunities on your behalf. If you're not signed to a label, you can work with your publisher and your performing-rights organization to seek out occasions to get your songs heard by music supervisors. When sending your demos to music publishers, be sure to emphasize that you have an interest in film and television and don't forget to point out any film or TV credits you might already have.

If you are a songwriter or a producer, you might need to learn the technical side of how to write for film and television. Consider taking courses in film composing from a local college or film school; they will help you develop your skills and your understanding of working within the film and television industries. Furthermore, those courses can generate valuable leads into the business.

Assemble a demo that shows your work in synchronization with visual images. Try to get involved with as many visual projects as possible, remembering that you may have to initially trade cash for credits. By making yourself available as a composer or a producer

for independent projects, student films, or community-access television on a low-cost, spec, or deferred-fee basis, you can build your résumé and possibly get in on the ground floor of the next major director's career. As with all networking in the business, the more you make yourself available, the faster your contact list will grow.

Film and television trade magazines (such as *Variety* or the *Hollywood Reporter*) can provide leads to music supervisors, music-library companies, and projects that need songs or styles of music that you are adept at producing. Directories such as the Music Business Registry (www.musicregistry.com) can give you names of music supervisors for your list of contacts.

James Leach, SESAC's director of writer-publisher relations in the Los Angeles office, is a veteran in the film and television field. He gives this advice to composers and producers trying to break into film and television: "Most of all, be the best musician you can be. The great music people in the business can weave the texture of their work into the visual. If you have a broad sense of music—being able to cover a variety of styles and actually compose and play music instead of simply program a synthesizer—your chances are much better. Have great people skills and don't be afraid to hustle."

FINAL CUT

Whether you want to shift your career away from recording and performing or you want to diversify your musical assets, music for film and television continues to be a growing opportunity. If you plan carefully and look at the screen from a different point of view, you can expect to hear your songs and see your name as the credits roll.

Michael A. Aczon *is the music supervisor for* Alcatraz Is Not an Island, *a documentary about the occupation of Alcatraz Island by Native Americans. He would like to thank music supervisor and publisher Ronda Espy for her assistance with this article.*

REVIEWS

M O T U

DIGITAL PERFORMER 3.01 (MAC)

A venerable audio sequencer gets a power boost and a jazzy new look.

By Rob Shrock

Mark of the Unicorn's (MOTU's) *Digital Performer* has been a first-call digital sequencer for almost as long as MIDI has been around—and for good reason. *Digital Performer* has maintained a healthy balance of power, stability, productivity, integration, and ease of use. Version 3.01 of *Digital Performer* continues the tradition by adding some significant new features, including dual-processor support, a powerful new Sequence Editor window, extensive surround-sound capabilities, better graphic editing of MIDI data, support for importing and exporting *Pro Tools* files, 14 new native plug-ins, and several notation enhancements. *Digital Performer* now has a newly designed user interface with a look and feel based on Apple's Aqua graphic design for Mac OS X.

Digital Performer 3.01 is a complete sequencing, recording, editing, mixing, and mastering environment. When combined with MOTU's USB-based MIDI Timepiece AV, MIDI timing is significantly improved through the company's proprietary MIDI Time Stamping (MTS) technology. The number of MIDI and mono, stereo, or surround audio tracks *Digital Performer* offers is limited only by your computer's processing power. The program supports audio formats with resolutions as high

FIG. 1: *Digital Performer*'s Tracks window displays track names, MIDI and audio I/O assignments, track colors, takes, and other pertinent data. The Transport window now sports three slide-out panels (top right) that provide instant access to hardware, synchronization, and editing windows.

as 24 bits and sampling rates as high as 96 kHz. In addition to MOTU's extensive line of audio interfaces, *Digital Performer* supports TDM, Direct I/O, ASIO, and Sound Manager hardware.

TRACKS AND TRANSPORT

Digital Performer's basic features have been covered in previous **EM** reviews (see the October 2000 issue for a review of version 2.7). The software boasts an extensive feature set for recording and mixing MIDI and audio; discussing everything in a single review would be impossible, so I'll stick to the highlights and concentrate on the new features.

The Tracks window (see **Fig. 1**) displays MIDI and audio-track content, MIDI and audio I/O assignments, patch assignments, and other related parameters. Each track (audio or MIDI) can contain multiple Takes, and new Takes can be created automatically if overdubbing in Cycle mode. That is a great way to build up a collection of performances without cluttering up the desktop with numerous tracks.

The Control panel offers tape-deck-style transport controls and metronome, overdub, loop, count-off, and sync controls. Three new slide-out panels provide handy access to audio-hardware settings, tempo-related parameters, and buttons for accessing the editing windows. You can rearrange the panels so that only one or two are showing, to save screen space.

SEQUENCE SAVVY

The new Sequence Editor window is one of the most significant additions to *Digital Performer*. The Sequence Editor window is like the Tracks window on steroids; now, you can graphically edit MIDI and audio simultaneously (see **Fig. 2**). The Sequence Editor is highly customizable: as many or as few tracks as you want can be displayed;

THIRD-PARTY ON!

In addition to the great plug-ins native to *Digital Performer* 3.01, many third-party developers directly support MOTU Audio System (MAS), the core audio engine of *Digital Performer* (see **Fig. A**). The following list is not exhaustive, but it includes the *Digital Performer*–compatible plug-ins that are essential tools for serious, professional work.

Audio Ease recently released *Altiverb*, a groundbreaking reverb plug-in that models actual acoustic environments by using impulse response samples (similar to the Sony DRE-S77 hardware reverb). Audio Ease also makes the Rocket Science and Nautilus bundles of quirky but fun and useful plug-ins.

Metric Halo's *SpectraFoo* provides comprehensive metering software, including spectrum analyzers, spectrographs, code and matrix meters, and a host of other detailed displays. Metric Halo also makes *ChannelStrip*, a high-end software console channel with 6-band parametric EQ, gate and compression with sidechains, and excellent metering.

Bomb Factory makes some of the finest hardware-emulation plug-ins, including accurate models of the Teletronix LA-2A, the Universal Audio 1176, the Pultec EQP-1A, the Fairchild 660, and the Tech 21 SansAmp (a model

of a modeler!), as well as several Joemeek and Moogerfooger products.

Waves Gold Native bundle is a professional's team of virtual workhorses. It includes analog- and digital-style EQs and compressors; world-class mastering processors; wonderful reverbs; and excellent delay-based, frequency, and pitch effects. Antares's one-of-a-kind *Microphone Modeler* is a great tool for creative coloration of miked signals, and its legendary *Auto-Tune* plug-in is a must-have in the world of pop-music production. Kind of Loud *RealVerb* is one of the best-sounding and most versatile native reverb plug-ins available. TC Works TC Native Bundle and DUY EverPack both include great EQ, compression, reverb, and noise-reduction plug-ins.

TC Works has recently shipped PowerCore, a digital signal processing accelerator PCI card designed to run TC PowerCore plug-ins. That allows *Digital Performer* to run *MegaReverb*, *TC MasterX*, and other high-end TC plug-ins previously available only for TDM. *Digital Performer* also offers full integration with Synchro Arts *VocAlign*,

FIG. A: Although *Digital Performer* comes packed with plenty of high-quality plug-ins, third-party developers have greatly expanded the pro-level processing options.

which provides state-of-the-art automated audio alignment for ADR and other synchronization tasks.

Although *Digital Performer* does not directly support VST plug-ins and VST Instruments, several applications can act as "shells" to allow MAS-compatible access to the world of VST. Audio Ease *VST Wrapper* is a no-nonsense MAS plug-in specifically dedicated to hosting VST plug-ins and VST Instruments. TC Works *Spark* opens up directly as a MAS plug-in, supplying a versatile routing matrix of four parallel streams of native and third-party VST processing. Cycling '74's *Pluggo* is arguably the best value going, providing more than 74 unique plug-ins in addition to serving as a VST plug-in portal.

each track can be individually resized vertically with Zoom buttons, drop-down minimenus, and shortcuts; audio and MIDI tracks can be armed for recording and scrubbed; and automation can easily be edited graphically. In fact, after setting up the basics of my sequences in the Tracks window, I mostly use the Sequence Editor as home base.

The Sequence Editor includes a new QuickTime Movie track that maintains frame-accurate video synchronization with the Conductor track. Individual movie frames appear along the same timeline as audio and MIDI data, and the guide frames intelligently adapt to the current zoom level without overlapping. The Sequence Editor window is a great tool for post scoring, especially if you can dedicate a second monitor to display the full-motion video on *Digital Performer*'s separate QuickTime Movie window (which scrubs frame by frame as you drag Soundbites, automation control points, and other onscreen elements). Even on a single monitor, after you have scored a few times with the QuickTime Movie window locked to the audio and MIDI tracks, you will want to toss your VCR remote control out the window.

GRAPHIC BITS
The Graphic Editing window (see **Fig. 3**) has been greatly improved in several areas and now includes some features reminiscent of Opcode's dearly departed *Studio Vision*. For example, you can view multiple MIDI tracks simultaneously, enabling you to edit, say, ten tracks of MIDI drums in a single window. Notes retain the color assignment of their respective tracks for easy editing.
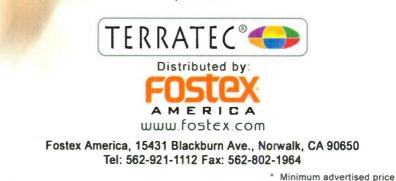
The Continuous Data Grid within the Graphic Editing window has also been redesigned to allow much more flexible viewing, filtering, and editing. Data can be viewed as points, bars, or lines; color

shading in Bars mode makes seeing the contour of the controller changes easier. Graphic editing of controller data is a breeze, even with multiple controllers superimposed on each other.

The versatile Pencil tool in *Digital Performer*'s new Tool Bar can draw in data points and lines in a multitude of ways. The shape or periodic waveform generated by the Pencil tool can be set to free, straight line, flat, triangle, sine, square, random lines, random steps,

parabola, or spline. A separate Reshape tool lets you redraw, scale, limit, or add to the current values. The graphic shapes that you create can snap to the edit grid to produce modulation effects in time with the tempo map. In short, some cool continuous controller effects— such as panning, volume changes, or filtering—can be penciled in quickly, which can make an exciting sequence out of mundane parts if you supply a little creativity.
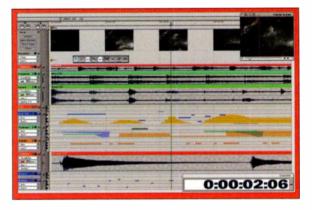
FIG. 2: *Digital Performer*'s new Sequence Editor window allows detailed graphic editing of MIDI and audio tracks. QuickTime movie frames appear along the audio timeline as well as in a dedicated QuickTime window. Individual tracks and their contents can be resized for better viewing and editing.

Zooming, looping, and scrubbing tools are also included in the Tool Bar, which you can flip between a horizontal and vertical orientation. Too bad it can't be reshaped into a 3-by-3 box, which would be my preference.

ALTERNATE AVENUES

Digital Performer's Event List remains largely unchanged; it displays in a chronological list almost every type of data that you might need to edit. You can filter the list so that you see only what you want. The Drum Editor window lets you view and edit notes along a timeline grid, and you can create composite drum kits from multiple MIDI devices. With the handy Rhythm Brush tool, you can quickly "paint" patterns into the window. The Drum Editor is great for programming grid-style "drum-machine" percussion parts.

The QuickScribe feature in *Digital Performer* has always been quite good at creating usable notation from MIDI tracks. QuickScribe has been updated in *Digital Performer* to include context-sensitive note spelling and dynamic hand splitting for two-staff piano parts. A Switch Staff command lets you jump notes to the opposite staff in piano parts. A collection of dynamics symbols, crescendos, and decrescendos can be added to the score, which you can now zoom to any size.

The Waveform Editor is as powerful as many standalone audio editors, al-lowing viewing and editing down to the single-sample level, and you can apply Premiere and MOTU Audio System (MAS) plug-ins to the audio. All edits in the Waveform Editor are destructive, however, so working on a copy is a good habit to form, especially because *Digital Performer* supports one undo level only. (MOTU plans to release version 3.1 by the time this review goes to print. In addition to providing direct waveform editing of surround-audio files, the free update to 3.01 adds unlimited undo capability with sequential as well as timeline histories. For more about the update, see the sidebar "Coming Attractions.")

Digital Performer includes excellent sampling-rate and bit-depth conversion processes. Both can be configured to provide a draft/fast mode operation in the foreground for immediate use while the high-quality process runs in the background, minimizing disturbances to the workflow. Although 16- and 24-bit files can't play back at the same time (they can exist in the same file), the high-quality conversion processes provide a practical means for combining files with different resolutions and sampling rates.

QUITE AN EFFECT

Digital Performer ships with 35 native 32-bit plug-ins, all of which support automation. In addition, many of the plug-ins are surround-sound capable, and new *Calibration* and *Bass Manager* plug-ins assist with surround setups. The included plug-ins sound great, and MOTU has provided a solid selection of useful tools for a wide range of projects. If your requirements are more specific or esoteric, however, most third-party plug-ins are compatible with *Digital Performer* (see the sidebar "Third-Party On!").

Some standouts among the native plug-ins are *PreAmp 1* (distortion and tonal shaping), *Sonic Modulator* (crazy modulation effects), *Multimode Filter* (a Moog-style filter and modulator), *Plate* (a simple yet great-sounding plate-reverb effect), *MasterWorks Limiter* (a surround-capable mastering limiter), *MasterWorks Gate* (one of the best gate plug-ins on the market), *Trigger* (providing audio in to MIDI Out), and *Trim* (for optimizing level, panning, and polarity).

Several virtual synths and software samplers support MAS directly, allowing the instruments' outputs to appear in *Digital Performer*'s instrument list. The programs include BitHeadz's *Unity AS-1* software sampler and *Retro AS-1* synthesizer; Native Instruments' *Reaktor*, *B4, Pro-52, Dynamo, Absynth, FM7*, and *Battery*; Koblo's *Studio 9000*; and IK Multimedia's *SampleTank*.

Propellerhead's *ReBirth RB-338* and *Reason* provide a virtual rack of synths, samplers, matrix sequencers, drum machines, and mixers by taking advantage of *Digital Performer*'s built-in ReWire compatibility. Audio Ease's *VST Wrapper* provides access to VST Instruments.

MIX ME A STRONG ONE

Digital Performer's Mixing Board window acts as a virtual mixer, with an automation section that rivals the best hardware consoles (see **Fig. 4**). Each track
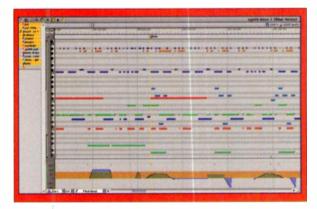
FIG. 3: In *Digital Performer*'s redesigned Graphic Editing window, you can view and edit several tracks at once. Each track appears in its assigned color.

can have as many as 20 inserts and 64 buses (32 stereo), and a track can be bused to a new channel strip for further processing if those aren't enough. You can create an unlimited number of custom Mixing Board configurations that can display any combination of audio, MIDI, aux, and master-fader tracks in the same console. A click of a button takes a snapshot of the mixer setup, and an unlimited number of fully automated mixes can be created and recalled instantly.

Control of the automation process comes through five automation modes that you can select from a drop-down menu in each channel strip. Once the automation data is written, you can edit it graphically or numerically. Almost everything on the Mixing Board can be automated, and a detailed Automation Setup window lets you customize automation parameters globally and for individual tracks. MOTU's native plug-ins support real-time automation, and many third-party MAS plug-ins currently or soon will support automation, as well.

My wish list for the Mixing Board window is short but directed: I would like to be able to open a console snapshot (levels, aux routings, master settings, effects, and so forth) in another sequence for projects that include multiple files with identical board layouts—a critical requirement for audiovisual work with a lot of similar cues. (Although you can use Clippings to export multiple effects bundles for individual tracks, setting up a new mixer from scratch is still time consuming.) Also, *Digital Performer* offers only four mono aux sends, which can easily get used up in complex mixes with stereo effects and detailed routing. (For some applications, I sometimes like to feed compression and EQ from sends rather than from inserts.) Eight sends would be better; a user-definable number would be ideal.

In spite of those minor gripes, however, I love the mixer in *Digital Performer*

COMING ATTRACTIONS

By the time this review goes to press, Mark of the Unicorn plans to release a significant upgrade to *Digital Performer* 3.01. According to the company, version 3.1 will add several new enhancements and features. Following are some noteworthy items:

General Enhancements

Unlimited multiple undo and redo with an Undo History window, time stamps on each undo event, an undo timeline, and branching undo histories

Complete destructive waveform editing of surround audio files

Edge editing of multiple Soundbites in one operation

Pro Tools–style insertion cursor and selection shortcuts

Enhanced batch importing of audio

Post-Production Features

Global time format display (for example, display SMPTE time code throughout *Digital Performer* with one easy setting)

Edit grid support for any time format (SMPTE frames, milliseconds, samples)

Digital-video playback using FireWire (no video card needed)

Music/MIDI Production Features

Recycle (REX) file import using drag and drop or import and audition

Tap Tempo entry (set tempos anywhere in *Digital Performer* by tapping, even during playback)

Enhanced Quantize (quantize almost anything, including markers, controllers, patch changes, audio mix-automation data, and so forth)

Paste Repeat (fills the selected region with consecutive pastes)

Real-time MIDI note Velocity expansion and compression

FIG. 4: Combining MIDI and audio tracks in the same display, *Digital Performer*'s sophisticated Mixing Board window lets you recall multiple mixes containing elaborate automation settings and effects processing.

and prefer it hands down to any other virtual mixer, including the one in Digidesign's *Pro Tools*. The Mixing Board window in *Digital Performer* is elegant, clean, and easy to use, yet it's extremely powerful. Coupled with a good hardware controller such as Mackie's HUI, Radikal Technologies' SAC-2K (see p. 146 for a review), or JLCooper's MCS-3800, the mixer in *Digital Performer* is capable of handling virtually any project of any size. With the extensive surround capabilities in *Digital Performer* and the current crop of rocket-ship dual-processor Mac G4s, the movement toward native processing and *Digital Performer* that is occurring in professional A/V-post facilities and music-production studios will continue to grow.

SURROUND THE TOWN

As you might expect, *Digital Performer* 3.01 does 5.1 surround in a big way. In fact, it also supports Quad, LCRS, 6.1, 7.1, and 10.2 multichannel audio. As new multichannel formats emerge, the program is already prepared to accommodate the routing through its open-architecture design. A lot of thought regarding proper surround-sound design has gone into the software, and it shows. MOTU has really done it right.

Each track in *Digital Performer* can have its own independent native or third-party surround panner. The program includes four advanced surround panners: *ArcPanner* (a high-resolution

panning dish), *n-Panner* (providing Cartesian coordinates on a square grid), *TriPan* (offering sophisticated three-way divergence controls), and *Auralizer* (a room simulator that uses psychoacoustic cues for localization). Several panning modes (Mirror, Parallel, Asymmetric, Balance, and Mono) assist in manipulating stereo tracks within a surround matrix. *Digital Performer* supports Mac OS Input Sprockets, so you can use a USB-compatible joystick to control surround panning.

Signals in *Digital Performer* are now routed through the Audio Bundles window (see **Fig. 5**). That window provides a matrix that lets you combine groups of inputs, outputs, and buses that mirror your audio hardware interfaces and the virtual buses in the Mixing Board window. With just a few mouse-clicks, you can easily create multiple output configurations (including surround setups) for your various routing requirements and recall them instantly. For example, you can fold down the multiple outputs of a surround setup into a stereo mix pair by loading a different Audio Bundle configuration. More elaborate and creative setups are possible if you have a good supply of hardware I/O and a bit of imagination. I was able to quickly and painlessly get a 5.1-surround configuration going while retaining my normal 2-track setup.

The implementation of Audio Bundles now requires you to open a new bundle in the Tracks or Sequence Edit window the first time you activate an input or output. That makes sense, because a fully loaded bundle could contain a lot of I/O that you might not want to see all the time while recording.

MAKE IT EASY ON YOURSELF

Digital Performer also hosts a collection of new convenience-oriented features. In Continuous Scrolling mode, the

playback wiper stays positioned in the middle of the screen while the data scrolls behind. (The Moving Wiper mode is still available.) The wiper now serves as an anchor when zooming in and out, maintaining the focal point when you're editing, and saving you from having to scroll to regain your location, as in previous versions.

Digital Performer also lets you execute the Save command while a sequence is playing (although doing so when a dense sequence of audio and MIDI is playing can result in momentary freezes in MIDI tracks). A custom key-binding feature allows almost every operation in *Digital Performer* to have a customizable key command that is saved in a separate file. The program even has key-binding sets that replicate the keyboard shortcuts in *Pro Tools*, Emagic's *Logic Audio*, Steinberg's *Cubase VST*, and *Studio Vision Pro*.

PRODUCT SUMMARY

Mark of the Unicorn

Digital Performer 3.01 (Mac)
digital audio sequencer
$795
Upgrade from AudioDesk or competitor
$395
Upgrade from Digital Performer
$295

FEATURES	4.5
EASE OF USE	4.5
DOCUMENTATION	4.5
VALUE	5.0

RATING PRODUCTS FROM 1 TO 5

PROS: Dual-processor support. New Sequence Editor window. Powerful re-design of MIDI graphic editing. Excellent surround-sound support. World-class mixer and automation. Superb native plug-ins.

CONS: No separate Volume or Velocity control for individual Soundbites. Only four aux sends per channel in mixer. Can't play back 16- and 24-bit audio files in the same song.

Manufacturer

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tel. (617) 576-2760
e-mail info@motu.com
Web www.motu.com

FIG. 5: *Digital Performer*'s Audio Bundles window lets you create complex hardware routing setups that you can store and recall as needed.

Digital Performer can now import as well as export Open Media Framework (OMF) files, allowing projects to be shuttled back and forth between Pro Tools and Paris systems. Entire mixes are not transferred intact, but the basic audio locations, MIDI data, fader levels, and a few other parameters translate perfectly. I've used that method many times on projects that were produced in *Digital Performer* and later mixed in *Pro Tools*.

Contrary to rumor, *Digital Performer* easily handles 24-bit OMF transfers in both directions; *Pro Tools*, however, requires an additional Digidesign application called *DigiTranslator* to complete the handshake with 24-bit audio. A 16-bit OMF transfer does not require any additional software.

Speaking of *Pro Tools*, I was curious to see how the mixing capabilities in *Digital Performer* would compare to my past experiences with *Pro Tools* now that I've upgraded to a dual-processor Macintosh G4/800 MHz. As a test, I set up a mix with 14 stereo tracks and 10 mono tracks and processed the various tracks with an assortment of plug-ins. I added eight stereo native MOTU *Chorus* plug-ins; eight stereo and five mono 4-band *Renaissance EQ*s and four mono *Renaissance Compressor*s from Waves; eight stereo Metric Halo *ChannelStrip*s; four stereo *Pultec EQP-1A*s and *LA-2A*s and two mono Bomb Factory *1176*s; one KOL *RealVerb*; and one stereo Audio Ease *Altiverb* in No-Latency (high-processor-load) mode before I started to hit red in the processor windows. I did not come close to taxing *Digital Performer*'s play buffers, which means that I could

have easily doubled or maybe even tripled the track count.

My test setup equaled 38 audio tracks with 16 tracks of chorus, 188 bands of EQ, 30 bands of compression, 2 huge great-sounding reverbs, and 8 channels of gating. That was without submixing or printing anything, which I would typically do, especially with big reverbs. Also, keep in mind that I was using some excellent high-end plug-ins that are known to be heavy processor draws, especially *Altiverb*. Considering that I could have doubled my track count, it seems that I can now rival or maybe even surpass my experiences with average Pro Tools systems. My little test may not have been very scientific, but it was nevertheless an eye-opener.

▼

The Waveform Editor is as powerful as many standalone audio editors.

WISH LIST

Overall, *Digital Performer* is stable. It consumes noticeably more CPU power and RAM than version 2.7, so seriously consider taking advantage of the new dual-processor Macs if your recording and mixing demands are great.

Most of my wish list is pretty nitpicky, though there are a few significant features I'd like to see. For example, audio editing would be easier if punching in over an existing Soundbite would graphically truncate the remainder of the original Soundbite rather than merely overlap it. If you punch in over a Soundbite, the edge of the previous Soundbite is not cut off at the punch-in point by the start of the new Soundbite. If you were to drag the new (punched-in) Soundbite to another track, you would still see the remaining part of the original Soundbite. The

remaining Soundbite fragments can accumulate and add confusion to the editing process. In addition, individual Soundbites sorely need their own level controls or Velocity settings, which would make dialog or track comping easier by providing a way to smooth out levels before applying volume automation.

Clippings (which are unchanged from the previous version) could be improved by having them include the Track names and audio or MIDI assignments. That would let complex templates be added to an existing sequence. Finally, the 946-page *User Guide* and the 155-page *Getting Started* manual are well written and nicely illustrated, but the 18-page *User Guide* index omits several key features, which are therefore difficult to locate in the text.

WINNING COMBINATION

Digital Performer is a rock-solid winner. You would be hard-pressed to find anything in the world of audio production that you could not do with *Digital Performer*. It's impossible here to delineate all of the program's great features. The Sequence Editor and graphic controller editing options are welcome refinements to the MIDI and audio-editing processes; MOTU seems to be listening to its customers' requests. The new graphical user interface is elegant and reflective of the direction in which Apple is moving with Mac OS X.

The surround-sound capabilities, QuickTime video features, and unbelievably great automated mixer will ensure the proliferation of *Digital Performer* rigs in post facilities and studios that are moving to native systems as computing power rapidly increases. Coupled with MOTU's excellent audio and MIDI hardware interfaces, the entire system is truly formidable.

*Producer and keyboardist **Rob Shrock** has recorded or performed with Burt Bacharach, Garth Brooks, Ray Charles, Elvis Costello, Faith Hill, Whitney Houston, 'N Sync, Dionne Warwick, Stevie Wonder, and others.*

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KORG

RØDE

NTK

Finally, a top-notch tube condenser mic for less than a grand.

By Rob Shrock

One of the few things about recording that hasn't changed in the past decade is that you still need great microphones to make great recordings. Great mics often make the difference between compelling, sonically superior tracks and inferior or uninteresting ones. That is especially true when recording that most elusive of instruments, the human voice. But there's a catch: great microphones tend be expensive.

Thankfully, one thing about recording that *has* changed in the past decade is the ratio of price to quality in studio gear. The trend is lower prices for better (or at least comparable) quality, and large-diaphragm studio condenser mics have been at the forefront. But are any of those affordable microphones really as good as the more expensive ones? Perhaps. One thing's for sure: not all of them are.

Røde Microphones' latest creation is the NTK, an affordable large-diaphragm tube mic with Class A tube circuitry and a fixed cardioid polar pattern. The NTK ships in a cardboard carrying case with a power supply, a 30-foot cable for connecting the power supply to the mic, and the M2 standmount. The mic comes ensconced in a padded vinyl zipper pouch. The SM2 shockmount ($99) is optional.

BEAUTY AND BRAWN

Nothing about the NTK is flimsy. The components—mic, power supply, and cable—are sturdily constructed and appear to be built to last. The microphone is heavy; when using a boom, I had to weight the base of a tripod mic stand to prevent it from tipping over. It's evidently rugged, too: I accidentally clobbered my forehead with the mic while positioning it on a stand. No harm came to the NTK, but it did leave a nice bump above my eye.

The NTK looks great. The mic's cylindrical body and grille basket boast a satin-nickel finish reminiscent of classic microphones from the past An inset brass dot just below the grille—a Røde signature—identifies the capsule's address side.

Attaching the microphone to the M2 standmount is easy and foolproof: simply unscrew a sturdy plate at the base of the mic, insert the mic base through the mount, and screw the plate back on. The mic couples to the optional shockmount in the same manner. Either mount holds the NTK firmly in position.

The microphone connects to the external power supply through a proprietary 7-pin cable rather than a standard 3-pin XLR cable. Many other tube mics also employ special cables, so that's nothing new. However, care must be taken not to damage or lose the 7-pin cable, or the mic will not be operational. (Røde claims quick turnaround in the event that you need an additional cable.)

A cool blue LED glows on the front of the NTK's power supply when the unit is on. The power supply can be externally toggled between 110V and 220V operation using a recessed switch, which supplies the necessary power to the mic and eliminates the need for phantom power. A slow-blow fuse resides inside the IEC power-connector socket. A standard 3-pin XLR connector provides signal output from the rear panel of the power supply.

PLEASING CURVES

Fine craftsmanship aside, it was the sound of the NTK that really got me going. I had my first opportunity to use the mic on recently signed Arista/Nashville recording artist Kristy Lee. I had produced Lee's original demos at my personal studio and used a Neumann KM 86 for her previous vocal tracks, pleasing everyone with the results. After her label signing, Lee came back to record a few of my songs for the label, and that happened to be during the week that a pair of NTKs arrived for review.

Experience has taught me that certain microphone and voice combinations work well and that others don't, no matter how good the voice or the mic. The best thing is to try out combinations of mics and preamps until you find a signal chain that best complements the vocalist. I already knew the KM 86 flattered Lee's voice, so before

The Røde NTK proves that affordable doesn't necessarily mean second-rate. Remarkably quiet, rugged, capable of high SPL handling, and downright yummy sounding—what more do you want for less than a thousand bucks?

the session started, I set up the KM 86, a Neumann TLM 103, and one NTK. I ran the three mics through the same signal chain: a Neve 1272 preamp into an Empirical Labs Distressor (set to Opto mode) and from there into a Mark of the Unicorn (MOTU) 1296 audio interface.

I soon eliminated the TLM 103—it didn't sound bad, just too neutral and polite. Although the TLM 103 sounded more up-front and personal than the KM 86, the KM 86 had more character. But when I tried the NTK, I knew after only two bars that it was the mic for the day. It had not only the up-front quality that I liked in the TLM 103 (both are large-diaphragm condensers) but also the personality of the KM 86.

Clearly, the NTK was designed to be musically pleasing rather than clinically accurate. It has a gentle presence boost from about 2 to 5 kHz and another slightly bigger boost centered at about 12 kHz. The boosts opened up the sound at exactly the right places to complement Lee's voice. In fact, Lee likes the vocal tracks so much that she wants the label to use them for the final record.

GOOD PERSONALITY

I really like the NTK's personality. The mic is not linear in its response; rather, the tonal color blossoms as the source gets louder. That makes the NTK sound just right for many sources, especially lead vocals. It also requires that you take more care with mic placement. The reward, though, is added tonal control.

For example, having the vocalist sing really loudly close to the mic will create a fuller and more aggressive sound than when you have the singer step back from the mic: the tone becomes more neutral as the vocalist backs away. Even with the levels matched, the two tracks will sound different—and not just because of bass boosting from the proximity effect. (The effect is roughly analogous to the difference in sound you get from turning down the volume knob on an electric guitar as opposed to turning down the amp.) On really loud, belting vocals, I preferred to keep the singer closer to the microphone to maintain that aggressive edge and then simply to back off the preamp. I loved that sound.

TWIN PEAKS

Although NTKs are not sold as matched pairs per se, Røde claims that any two NTKs will perform almost identically, thanks to tightly monitored quality control and the fact that the twin-triode 6922-model tubes are stringently hand-selected. That claim was borne out in my testing.

I'm presently producing tracks for a set of female twins called Coppola, and the NTKs worked well for their voices, too. I tried a Neumann U 47 and U 67 on them first, which sounded great;

however, the tonal difference between those two mics was too drastic—the twins' voices are very similar in nature, and I wanted to preserve that element. As it turned out, I was just as pleased with the sound of the NTKs on their voices. For less than the price of one vintage tube mic, I had two tube mics that I liked just as well.

The twins record their vocals at the same time while standing next to each other. The NTKs are fixed in a cardioid pattern, so I needed to experiment to find the mic positions that afforded the best rejection and phase coherency. As I discovered, the NTK provides a good bit of off-axis rejection, especially at 90 and 270 degrees, so it's a viable option whenever side or rear rejection is important.

The NTK can take a lot of level, too. At one point, I thought I heard distortion in the NTKs—the twins can sing extremely loudly at times—but later I realized I was overdriving the preamps. Although the NTK does not offer an attenuation pad, Røde claims the mic can take levels as high as 158 dB, making it usable even for drum overheads and guitar amps. Later tests showed that to be true.

One potentially negative thing about the NTK is its tendency to accentuate sibilance from some sources. For example, the presence boost that worked so well on Lee's voice (and on other instruments) accentuated sibilance from the Coppola twins. Fortunately, I

solved the problem easily with a de-esser (I used Waves *DeEsser* plug-in). But in the end, the NTK's tonal character was well worth having to deal with a bit of sibilance.

IVORY TOWERS
Piano is my main instrument, so I was curious to check out the pair of NTKs

NTK Specifications

Acoustic Operating Principle	pressure-gradient transducer
Diaphragm	1", 6-micron-thick gold-vapor-deposited Mylar
Tube	twin-triode 6922
Polar Pattern	cardioid (fixed)
Frequency Response	20 Hz–20 kHz (±6 dB)
Dynamic Range	>147 dBA (for THD <1%)
Signal-to-Noise Ratio	>82 dBA
Self-Noise	<12 dBA
Maximum SPL	>158 dB (for 5% THD @ 1 kHz)
Dimensions	8.50" (H) × 2.28" (D)
Weight	1.68 lb.

on acoustic piano. The vaulted ceiling in my living room makes my Yamaha grand sound pretty darned good, so I set up the NTKs and tracked them through a MOTU 828 computer audio interface in to my PowerBook at 24-bit resolution.

The resulting tracks sounded excellent. Again, the NTKs' frequency boosts gave the piano a finished sound that didn't scream for EQ after the fact. The overall sound was big, but even the low-level material sounded great. The NTK is quiet, particularly for a tube microphone, so you can crank up the preamp to capture low-level material yet still acquire a clean, rich signal.

GUITARS AND TAMBOURINES

The NTK also sounded good on acoustic steel-string guitar. The tone wasn't particularly pure, though, meaning the NTK is probably better suited for rock tracks than, say, clean bluegrass guitar tones. The NTK nicely enhanced the aggressive edge of a hard-strummed part I recorded, providing a dose of attitude that fit the track well.

The pair of NTKs also worked well on a solo nylon-string guitar recorded in stereo. Again, the sound was not as pure as that provided by some mics. But the NTK's quietness lets you get a tube sound on tender acoustic-guitar passages, without the usual accompanying noise.

I also tested the NTK on triangle, tambourine, and egg shaker. Large-diaphragm tube condenser mics are typically not ideal for high-pitched percussion, and the NTK is no exception. Actually, the egg-shaker track didn't sound bad, but the tambourine and triangle tracks exhibited some distorted upper harmonics, resulting in an unfocused yet edgy sound that I didn't care for. That is not necessarily a deficiency on the NTK's part, but a reminder that it, like most mics, is not the perfect choice for every sound source.

MORE FOR THE MONEY

Although the old maxim "You get what you pay for" generally holds true, occasionally a product comes along that gives you more than you expect for your dollars. The Røde NTK large-diaphragm tube condenser mic is such a product. Despite being priced at less than a grand, the NTK will put you on equal footing, or better, with many new and vintage microphones that cost several times as much.

I love the NTK. Like the best of tube microphones, it has a sound—call it character, attitude, or what have you—that is musical to the ears. The mic's gentle presence boost makes just about everything sound good, especially vocals, and the tone really blossoms as you hit the capsule with more level. The NTK not only sounds great but also is quiet and can handle a lot of level. Moreover, the NTK is clearly built to withstand a lot of use. My guess is that it will still sound great 40 years from now. ✪

D R A W M E R

DS501 POWER GATE

Get as precise or creative as you want with this full-featured gate.

By Myles Boisen

A noise gate, though not the most glamorous of components, is an essential part of any well-equipped multitrack studio. By reducing noise, including leakage from headphones and neighboring instruments, the gate plays an important behind-the-scenes role, helping maintain the spaciousness and background clarity of a mix. In particular, live ensemble recordings, amplified instrument tracks, and drum tracks can benefit greatly from careful gating.

If you've had your fill of bare-bones gates that cause as many audio problems as they solve, check out the DS501 Power Gate from Drawmer, the company that has pioneered gating technology since the early 1980s.

PEARLY GATES

A sophisticated dual-channel noise gate, the DS501 is sized to fit into a single-rackspace box (see **Fig. 1**). Its two channels can be linked in stereo or used on two separate mono sources. Left and right channel controls are completely independent and are laid out side by side on their respective halves of the front panel. The Stereo Link switch is located in the middle of the unit, and the Power switch is on the far right.

Drawmer incorporated several features not commonly found on noise

gates into the DS501, but I'll discuss the unit's more conventional parameters first. The DS501's five continuously adjustable rotary knobs provide control over all aspects of a gated signal's envelope and dynamics. Those are, from left to right: Thresh (threshold), Attack (attack time), Hold (the length of time the gate is held open after the signal drops below threshold), Decay (the rate at which the gate closes after the signal drops below threshold), and Range (the amount of attenuation applied to the signal once the gate closes). The Range control remains active on each channel when the unit is in Stereo-Link mode.

Those parameters should be familiar to anyone who has used gates before; however, finding all of them on one unit is rare. A vertical four-LED ladder meter next to the Threshold control indicates signal strength relative to the selected threshold; a red light at the top of the display corresponds to above-threshold or gate-open status. Three LEDs across the top of the Gate section indicate gate status: red shows when the gate is closed, yellow indicates hold status, and green confirms that the gate is open.

A two-position Function switch selects Gate or Duck mode. Ducking is a common voice-over and broadcast procedure whereby a key source (a narrator's voice, for example) is used to automatically attenuate a music bed or other track. For that function, the Range knob governs the ducked audio's gain reduction when the vocal or other key input is present. Rounding out the Power Gate's feature set is an Output switch with three positions: bypass, gate, and key listen.

SHARP CORNERS

Typically, a noise gate opens and closes in response to the voltage of the audio signal that it's gating, but an external signal

connected to a key-input jack can also trigger the gate's detection circuitry. The key input is usually the same audio track but processed through outboard EQ to boost or eliminate certain frequencies. When gating a kick drum or floor tom, for example, it may be advantageous to remove some high frequencies so snare and cymbal sounds are less likely to trigger the opening of the gate. Likewise, boosting those drums' resonant low frequencies will tend to hold a gate open longer, producing a more natural decay characteristic.

The DS501 is unique in that it allows onboard equalization of internal or external key signals. That function, one of the unit's extra features, is handled in the Key Source section, which is located to the left of the conventional gate controls for each channel. The Key Source section provides a two-position switch

FIG. 1: The Drawmer DS501 Power Gate made a believer out of the author. The sophisticated dual-channel unit packs a wealth of useful features into a 1U box, providing remarkably precise control of signals.

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labeled Ext/Int and two continuously variable knobs for setting low-frequency and high-frequency filters. The two shelving-type filters sharply reduce frequencies below the selected low corner frequency and above the selected high corner frequency.

Setting the Ext/Int switch to Int (internal) allows the type of key filtering described previously, but without the need for an outboard equalizer; switching it to Ext (external input) permits ducking and other specialized applications. A kick drum, for example, can be used as an external key input to trigger and tighten up a bass guitar's attack in the mix. You can also construct wild effects by keying from metronomes, CDs, vocals, or other sources to gate or duck selected tracks. In addition, you can route the key source and any associated EQ processing to the DS501's outputs simply by switching the Output switch to the key-listen position—a handy feature that lets you hear exactly what you're keying with.

Another Power Gate innovation called Peak Punch allows for tunable transient boosting. Located to the right of the gating controls, the Peak Punch section provides a three-position switch and two continuously variable knobs marked Tune and Level (see **Fig. 2**).

The three modes accessed by the switch are Bypass (marked "Out" on the front panel), Full-Band (marked simply "Full"), and Frequency-Tunable (marked with an EQ-boost graphic). In Full-Band mode, you can boost the signal as much as 12 dB using the Level knob (arbitrarily labeled 1 through 10). In Frequency-Tunable mode, you can apply as much as 12 dB of gain to a spe-

FIG. 2: The Power Gate's unique Peak Punch circuit lets the user selectively boost desirable transients of the gated signal, adding clarity and punch.

cific frequency range, which is determined by the Tune control. In both active modes, the Peak Punch circuit provides a 10 ms gain boost every time the gate opens, reinforcing the initial transient of the gated audio.

ON THE FLIP SIDE
The DS501's rear panel provides +4 dBu balanced XLR connectors for input and output (see **Fig. 3**). There is no provision for –10 dBV signals, and unbalanced operation requires custom cable soldering. A 2-conductor ¼-inch jack on each channel accepts key-input signals. The unit has a standard IEC power-cord jack. Voltage selection and fusing are located inside the sturdy all-steel chassis and thus are not accessible without opening the unit.

Drawmer prepared a clearly written and comprehensive manual with helpful diagrams; five minutes with that exemplary manual were all I needed to decode the DS501's more esoteric features.

KICK START
Drums are typically the most difficult instruments to gate properly. While testing the Power Gate, I focused on a variety of challenging sources from a standard rock drum kit. Starting with kick drum, I was immediately impressed by the sensitivity and forgiving nature of the DS501's attack control. With proper adjustment, that feature allows a musical softening of the kick drum's attack without the audible clicking that severely lessens the usefulness of some gates. I also appreciated the continuously variable Range control. Typically, 20 to 30 dB of gain reduction is all I require for most gating applications, and having control over that parameter certainly contributed to a more natural sound.

The Power Gate's key filter is another powerful feature. It proved especially useful for tuning out snare sound on the kick track. But the key filter must be used in moderation in that application because the high frequencies of the beater attack are crucial for opening the gate and

the lows keep it open. When set higher than 3 in full-band position, the Peak Punch elicited an unpleasant *pock* from the kick track. In Frequency-Tunable mode, low-end boosting added an undesirable boxiness to the sound and was not effective for adding any real punch. Set above 1 kHz, however, the Peak Punch was handy for boosting the click of the beater and helping the attack to cut through. Precise boosting in that fashion should work well to enhance kick-drum sounds in a range of musical styles.

It took me two passes through a standard three-minute song to dial in a tight, perfectly controlled kick-drum sound. The Bypass switch allowed a quick A/B comparison of the gate's comprehensive and precise control. Once adjusted properly, the bypass contributed no significant coloration; all that was missing from the kick track was snare-drum, cymbal, and bass-amp leakage. Switching between Bypass and Gate modes produced no clicks or muting of the audio, a point that further underscores the DS501's professional quality.

TOM-TOM CLUB
Another test track presented a challenge: a rack tom had been hit very lightly, which typically makes for difficulty in setting a reliable gate without getting a lot of snare bleed. The Power Gate's key filtering took charge of the situation by letting me lower the high-frequency shelving EQ and tune out a lot of the snare sound. In addition, a low-frequency cut seemed to tighten up the rack-tom triggering, as did a slight lengthening in attack time from 10 µs to 0.1 ms. Finally, a light Peak Punch boost at 500 Hz gave the tom track more tone—a save that left me with the impression that this innovative module could help a dead-sounding drum have more life in a mix. You can also use Peak Punch to add a bit of gain and aggressive attack to a drum track, which is especially helpful for rock mixes because the toms always seem to need an extra boost.

I got similarly positive results with the Power Gate on floor tom, although

DS501 Power Gate Specifications

Inputs	(2) +4 dBu balanced XLR; (2) ¼" key
Outputs	(2) +4 dBu balanced XLR
Maximum Input Level	+17 dBu
Maximum Output Level	+17 dBu
Frequency Response	23 Hz–31 kHz
Distortion (@ 0 dB input)	<0.025% (100 Hz, 1 kHz, 10 kHz)
Noise (RMS)	–93 dB (22 Hz–22 kHz, unity gain, gate open)
Threshold Range	∞ to –72 dBFS
Attack Time Range	10 µs–1 sec.
Hold Time	10 ms–2.5 sec.
Decay Rate	5 ms–4 sec.
Range of Attenuation	–80 dB–0 dB
Low-Frequency Filter	25 Hz–3 kHz
High-Frequency Filter	250 Hz–30 kHz
Tunable Transient-Boost Circuit	10 ms boost; 0–12 dB gain (variable);
(Peak Punch)	75 Hz–16 kHz (variable peak filter)
Dimensions	1U × 7.9" (D)
Weight	6 lb.

a longer decay time (about 1.5 seconds) was necessary. Thanks to the Peak Punch's Frequency-Tunable mode, I was able to enhance punch at 200 Hz or sharpen the high-end attack around 3 to 4 kHz. The short (10 ms) duration of the Peak Punch circuitry let the drum cut through a dense mix without unwanted coloration or the type of onboard EQ boosts that make a dull floor tom sound papery or thin. Nice trick, Drawmer!

SNARE THE PRIZE

Snare-drum gating is something I rarely do; typically, I reserve it for situations in which tight gating is required to keep an extreme effect confined to only the snare. To test the Power Gate's mettle in that application, I chose a track titled "Dance Craze Time" that I recorded for the Casino Royale CD *Where's the Tiger?* (Double Play, 2001). On the song, drummer Wes Anderson's snare

work includes a variety of hits, from subtle rim shots to full-on power backbeats. Gating the track was a challenge, and initially I had a lot of trouble with a dull, "boinky" tonality from the snare. In addition, certain kick-drum hits kept triggering the gate. Then I realized the Peak Punch circuit was engaged and the Level control was set at 7.

Switching Peak Punch out made the track significantly more manageable. I employed some key filtering below 250 Hz, and careful adjustment made every rim shot audible, with bass-drum attacks sneaking through only occasionally. Using the high-frequency shelving, I also filtered out the hi-hat somewhat. Again, slightly reducing the attack (to a setting just under 0.1 ms) minimized false triggering and gave the snare a more natural sound. Finally, I reengaged the Peak Punch function and set the Tune control to 4 kHz and the Level control to 4, which gave the

drum a bit more edge in the mix. Thanks to the DS501, gating was no longer a dreary chore; it was actually creative and fun.

While testing for coloration added by the DS501 (with Peak Punch out), I detected a slight increase in highs and definition on the snare track when I switched in the gate. Technically, any kind of coloration in a device such as the DS501 should be considered a flaw; in this case, however, the subtle change was beneficial. Is it possible, after all, to have too much definition on a drum track?

In its Stereo-Link mode on drum-overhead tracks, the Power Gate also performed impressively on a light, skittery jazz-drum performance. Indeed, the unit seemed to anticipate every nuance. A longer decay time (one second or more) made the gate's processing practically undetectable.

Using the Power Gate's key filter, I achieved interesting gated-reverb effects by tuning the gate to open for kick drum or snare hits on ambient room mics. I've also been known to gate an entire drum set as an effect, and when stereo-linked for that application, the DS501's ample controls did not disappoint me.

I had fun using the Duck control to attenuate rather than pass audio whenever a drum was hit. By suppressing all rhythmic accents, I created an "anti-drum" track that let only cymbal decays and rests come through. Although it's not an effect I use often on drums, ducking is a powerful option in the Power Gate's toolbox. A more common musical use would be to automatically reduce the level of loud guitars under a lead vocal.

RUNNING THE GAMUT

I was not expecting gating miracles when I tested the Power Gate on a

FIG. 3: The Drawmer DS501 Power Gate's rear panel provides +4 dBu balanced XLR I/O and ¼-inch key inputs on each channel.

quiet, avant-garde vocalist in a live-ensemble setting. Key filtering, however, proved invaluable for homing in on the singer's range and tuning out other instruments in the room. When whispery passages dropped below threshold, setting the Range control to a gentle −10 dB made it possible to achieve some gating control without completely losing the source.

On saxophone and oboe, giving a little extra attention to the DS501's Attack, Hold, and Key Filter settings paid off quickly, allowing soft, airy notes through without clipping off too much of the initial breath sounds. An atmospheric guitar track, produced by stroking the instrument with a rubber mallet, also proved surprisingly "gateable."

On nonpercussive sources such as synthesizer washes and volume-pedal guitar swells, gating is often more trouble than it's worth. But once again, the Power Gate's smooth attack characteristics proved a joy to use. In fact, during all of my testing, the only instrument that didn't really benefit from a detour through the Power Gate was a softly bowed cello.

While using the DS501 daily on several sessions, I encountered no problems with headroom, distortion, or noise. Indeed, I ended up with only one minor complaint: when adjustments of a few milliseconds are crucial, the Power Gate's small knobs and tight, tiny legends are less than accommodating to the fingers and eyes. Every one of the Power Gate's 18 rotary controls and 10 switches is essential, however, and getting them all on the face of a 1U box amounts to a minor miracle. Considering the versatility and depth of control options, it is a small inconvenience to adjust to the DS501's somewhat cramped faceplate.
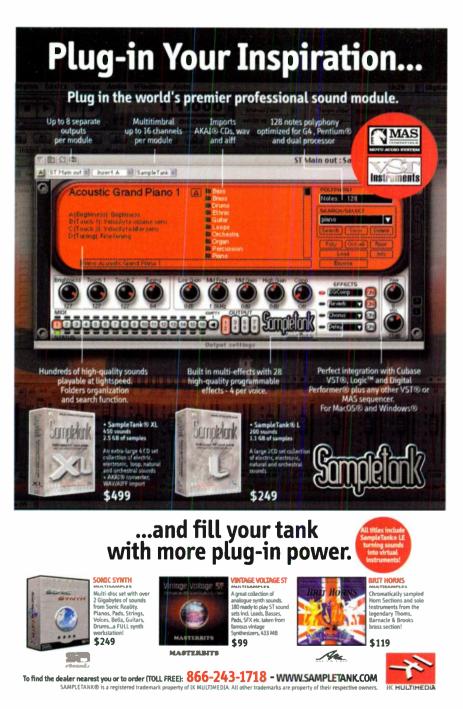
OPEN-AND-SHUT CASE

Before using the Drawmer DS501 Power Gate, I had regarded gates as a necessary evil. I never relished using them, largely because of their lack of parameter controls, but the Power Gate has changed that perception for good.

Aside from the unit's small knobs,

the only problem I had was figuring out how to afford eight Power Gates for my 16-track analog studio! The key-filtering feature makes so much sense that I can't imagine ever trying to gate a track without it. Although they are specialized functions, Duck and key listen expand the unit's appeal to include post-production and broadcast applications. The Peak Punch circuit, though best used in moderation, is a powerful problem solver for drums and other sources. The Power Gate is a well-built, professional tool that not only gets the job done right but also opens up a world of creative possibilities beyond the drab, utilitarian role usually assigned to noise gates.

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INA - GRM

GRM TOOLS RTAS 1.0.3 (MAC/WIN) AND GRM TOOLS TDM 1.3.2 (MAC/WIN)

A sound designer's dream is a mouse-click away.

By Alex Artaud

Look around today, and you'll find a slew of plug-in processors that can chew up your sounds and spit out gigabytes of bliss and mayhem. After checking out a bunch, it's easy to become a tad jaded because you're probably hearing dozens of effects that sound the same. That won't be the case with GRM Tools, an intelligent selection of plug-ins for Digidesign's *Pro Tools* software that will delight and amuse you. Developed by the Groupe de Recherches Musicales (GRM; part of the Institut National de l'Audiovisuel [INA] in Paris), GRM Tools was designed by composers for composers. **EM** first reviewed GRM Tools in the January 1997 issue, and in the interim, GRM added refinements that make the GRM Tools bundles indispensable for the serious sound designer.

GRM Tools TDM 1.3.2 and GRM Tools RTAS 1.0.3 are available for the Mac; a PC version should be available by the time

you read this. (A two-volume Mac and PC VST version is available from Steinberg.) The TDM and Real Time Audio-Suite (RTAS) versions aren't identical; see the sidebar, "RTAS: A Sleeker Look," to learn about the differences. I tested the program on a TDM system using a Mac G4/450 MHz with OS 9.1 running *Pro Tools* 5.1 with Pro Tools/24 Mixplus hardware. The TDM version requires a computer that will run *Pro Tools* software comfortably—I recommend nothing less than a PPC/400 MHz running OS 9.04.

GRM Tools includes plug-ins for comb and bandpass filtering, EQ, delay, Doppler effects, and three cryptic animals named *Pitch Accum, Shuffling*, and *Freezing*. I'll look at the general layout of GRM Tools TDM, each plug-in's main features, and a few application ideas.

GETTING THE GUI

Several GRM Tools plug-ins share the cursor-on-canvas approach of Arboretum's *Hyperprism*. Dragging the cursor, or Conducting Square (called the Conducting Ball in the RTAS version), controls at least two parameters at once. However, freely dragging the cursor around may not be ideal for controlling some plug-ins. By pressing the Control or Command key while dragging the mouse, you can limit the Conducting Square's movements to the horizontal or vertical axis. You can also reset the Conducting Square to a neutral location by holding down the Option key

and clicking on the Conducting Square. If you prefer, you can bypass the conducting window and use the mouse on a slider or button to increment values, or double-click on a field to type in values.

All plug-ins come with 16 preset buttons. (GRM nomenclature leaves something to be desired. A *preset* can be a set of parameters assigned to 1 of 16 buttons or an automated segment that morphs between two button settings.) Clicking on a button once instantly loads factory or user-configured settings. You can also use a slider to adjust the transition time between presets from instantaneous to as long as 30 seconds. Each plug-in also includes a Super Slider: a long horizontal slider at the bottom of the console that lets you interpolate between preset locations manually. Eight preset markers along the Super Slider can be assigned to any of the 16 preset locations.

Whether you are dragging the cursor through the active field or using the Super Slider to morph between presets, each plug-in can become a performance instrument. Because you can automate each plug-in's parameter sliders in *Pro*

FIG. 1: The GRM Tools *Comb Filter* plug-in offers a single-band or 5-band configuration. The master frequency, resonance, and lowpass sliders and the Super Slider are highlighted in green to indicate readiness for automation.

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Tools, it's possible to create unusual timbral shifts throughout a song.

COMB, BANDPASS, AND EQ IT

GRM Tools TDM provides two versions of *Comb Filter:* single band and five band (see **Fig. 1**). In the latter, you can adjust the frequency of the five bands between 45 Hz and 20 kHz and apply resonance and a lowpass filter to each band. Master controls act globally on the resonators. Although you can get recognizable flanging effects with the plug-in, it could also serve as an excellent tool for formant synthesis. That would allow you to morph the timbral signature of, say, a door slam into that of a chime. It's also good for boosting bass frequencies to give the low end more *oomph.* In addition, I liked it for subtle treatments on a drum kit.

The *Band Pass* plug-in has two separate windows, which lets you use the Conducting Square for discrete, left- or right-channel applications. *Band Pass* offers an adjustable center frequency and bandwidth and has the ability to bandreject either channel. Each window has sliders to control the lowpass and highpass cutoff frequencies.

Equalizer provides 23 fixed-frequency filters (40 Hz to 13 kHz) in a graphic EQ layout. Intervals are a third of an octave apart with 12 dB of boost and 100 dB of cut available. Although there is no Conducting Square to control this plug-in, you can click and drag the mouse across the faders as a quick way to draw a setting. You can also use the Super Slider to control all faders at once and even automate fader movements for playback. However, the radical boost and cut capabilities

FIG. 2: *Delay 24* provides as many as 24 delays within a specific range. You can accomplish classic delay effects or create bizarre treatments by adding random delay.

combined with the transparent quality of its sound make the plug-in perfect for any subtle EQ chores.

ENDLESS DELAY

Delay 24 provides a flexible environment for producing everything from classic delay to weird reverberant effects (see **Fig. 2**). Sounds can be repeated as many as 24 times with a maximum delay time of 683 ms. You specify how far apart successive delays will be and determine the amplitude of each successive delay.

As with any good delay unit, there's feedback control with the added convenience of cross-feedback control for stereo signals. Also, a random-delay feature includes a rate slider to control how often random numbers are generated. I used it to create odd shimmering effects on guitar and woodwinds. *Delay 24* works wonderfully for changing the attack characteristics of staccato sources, such as plucked stringed instruments, and giving them a slow, scraping attack.

DABBLING WITH DOPPLER

The *Doppler* plug-in lets you move sound around in space in real time by adjusting the speed, pitch, and stereo placement of the sound source. The Plug-In window shows the Conducting Square surrounded by a single small rectangular icon for mono signals or two icons for stereo signals (see **Fig. 3**). The icons circle the Conducting Square, giving you good visual reference for what's happening to the audio.

With *Doppler,* you can control how much the loudness and frequency change as the sound moves, and you can control the time the treated sound takes to catch up to the moving signal. Parameters such as Circle Frequency and Circle Amplitude let you control the amplitude of the sound's circular movement as well as how fast the sound rotates around the center. With stereo material,

FIG. 3: By shifting a sound in space, the *Doppler* plug-in can produce three-dimensional panning qualities.

you can put one channel's rotation out of phase with the other, creating asynchronous movement of sound between the speakers. Although *Doppler* lets you achieve classic Leslie effects, its strength is in bringing three-dimensional qualities to a sound's movement.

I automated the Super Slider on a droning string section so that the sound slowly moved between the speakers without the panning becoming a distraction. I then created a churning alien-machine effect by setting the frequency to 100 percent and gradually raising the amplitude. I also got good results by experimenting with the feedback slider on wind-chime and breaking-glass samples.

ACCUMULATE THAT PITCH

One parameter-heavy plug-in is *Pitch Accum,* a potentially unruly beast of a harmony processor. The plug-in offers two independent channels, each with transpose, delay, and gain sliders. The Conducting Square is "wired" to the transpose sliders in the plug-in and can shift them up and down by 24 semitones (two octaves). You can select periodic waveforms for frequency modulation (sine, triangle, sawtooth, square, and pulse), and you can add random modulation. A Feedback slider can enhance reverberant and arpeggiator effects, and a mono and stereo parameter directs signals to both channels simultaneously or independently. Also, a Direct slider is used to lower the source signal in the mix.

Using the Window slider, you can select the length of the signal fragment

FIG. 4: *Freezing* is one of the more unusual plug-ins in the set. It lets you capture audio fragments and mutate multiple loops into your signature performance.

being processed in real time. A Cross-fade slider controls how the original signal overlaps with the processed one, and a Pitch Detector algorithm adjusts the timing of the fragments according to the input signal's frequency.

Pitch Accum functions as much more than a harmonizer, though it certainly can do that job. For example, I used the plug-in to slowly transform a soprano voice into a group of male voices. I also easily warped the sound into a bubbling soup of frequencies and then back to the original soprano. In addition, *Pitch Accum* was useful as a phrasing algorithm. I used it to eliminate the melodic component of a passage and then used the Window and Cross-fade sliders to manipulate the sound's remaining rhythmic accents.

SHUFFLE AND FREEZE

Shuffling was one of my favorite plug-ins. It lets you scramble an input signal's fragments and splice them back together. You can control the fragments' lengths and how far apart they will be and also adjust their pitches. To thicken a sound, nudge the Density slider to the right and add feedback. For silences between fragments, move the Density slider to the left.

Shuffling is useful for creating crowd effects or sporadic bursts of sound that pan between speakers. I loved the effect of applying it to a looped, melodic singing voice. I set a fast attack and release time (0 percent on the envelope parameter) and adjusted the Fragment parameter to match the source loop. With some fine-tuning, I was able to create a percussive semblance of the mel-

RTAS: A SLEEKER LOOK

The GRM Tools TDM version has a decent user interface, but the RTAS version is a big improvement (see **Fig. A**). The RTAS plug-ins also provide operational advantages that TDM-only users may miss. For starters, RTAS users won't experience some TDM DSP limitations: the RTAS version offers as many as 128 delays in its *Delays* plug-in; the TDM version provides a maximum of 24. Also, most RTAS plug-ins cover a wider range of frequencies, and all have a Mix feature. The omission of Mix from the TDM version is a mystery.

The RTAS version is easier on the eyes. The darker control surface makes reading the parameters easier, as well as the added color in some of the plug-ins, such as *Doppler.*

GRM Tools RTAS is definitely a great value for host-based systems,

FIG. A: The RTAS plug-ins are easy to read and navigate. *Reson,* shown here, provides as many as 128 lowpass, highpass, and band-reject filters, which is way beyond the capabilities of the TDM version.

especially if you have a fast dual-processor Mac. Even if you do have Pro Tools hardware, the RTAS version may be a better way to go. Just be prepared for the plug-ins to grab your computer's processing resources in a big way.

ody that I mixed with the original loop.

Although *Shuffling* is great, the plug-in I really became hooked on was *Freezing*. That algorithm snatches and loops brief segments of audio (see **Fig. 4**). Sliders control the number of loops (two, four, six, and eight repetitions), change the timing of loop playback, modify loop speed and duration, and tweak loop synchronization. The captured loop can also be scrubbed using the Conducting Square.
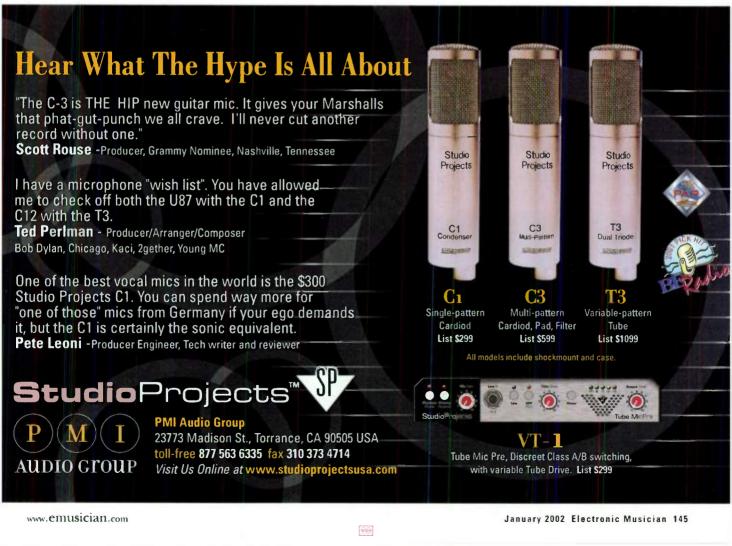
Freezing is the ultimate on-the-fly performance plug-in. It was especially useful on drum loops. I got a nice effect by duplicating a couple of drum tracks, dedicating a pair of tracks to the plug-in, and crossfading between dry and treated drums. I also used it to capture a quick bass drum and snare flurry and to play the selection with the Conducting Square and parameter sliders. That worked well at points in a song where some deconstruction seemed appropriate. Using the Conducting Square, I reduced the loop down to as little as 5 ms, scrubbed the segment to get flanging and tonal elements, and lengthened the loop to get recognizable snare hits. That also worked well with spoken and sung passages.

One drawback is that all GRM Tools plug-ins are quite DSP intensive. The largest segment I could freeze using my TDM system was 683 ms long. I'd have to use the RTAS version to get the three-second capture capability referred to in the manual. That is my only complaint

about GRM Tools, but I hope the manufacturer will address it in future versions.

SPIRIT OF SCHAEFFER

Like many musicians, I've combined analog synthesis and signal manipulation with the precision of digital recording and editing. I enjoy the rich sound and unpredictable nature of some analog approaches, even though I can't always replicate them. GRM Tools combines a flexible architecture with great-sounding algorithms that are never tedious. Their intelligent design connects them to the legacy of Pierre Schaeffer, whose spirit is still alive in the work of INA-GRM. But don't take this review as the final word. Download a two-week demo at the company's Web site, and you'll see what I mean. With the site's MP3 files and excellent support documentation, you'll be up and running in no time.

Alex Artaud is a musician, sound engineer, and writer living in Oakland, California.

RADIKAL

SAC-2K SOFTWARE ASSIGNED CONTROLLER

Lay your hands on this control surface.

By Brian Smithers

adikal's SAC-2K occupies a market niche somewhere between generic MIDI fader devices and high-end proprietary control surfaces. When used with a compliant host program, the SAC-2K provides transport controls, touch-sensitive moving faders, and an array of soft buttons and knobs that control anything from EQ settings to edit functions (see **Fig. 1**).

Despite its USB ports, the SAC-2K works only with a MIDI connection (see **Fig. 2**). Radikal has been working with Apple and Microsoft to straighten out unresolved issues related to USB's handling of timing data on Macs and PCs; a suitable USB driver is expected soon. Once those issues are settled, you'll be able to connect SAC-8X eight-fader expansion units to the SAC-2K's built-in USB hub. Radikal expects to release the expansion unit concurrently with USB support.

Installation of the SAC-2K is a complete no-brainer: On a Windows machine, you plug it in and set the software to find it according to the instructions in Radikal's manual. On a Macintosh, you need to configure the Open Music System (OMS) to recognize the SAC-2K. Even if you've never used OMS, you probably won't need the manual; if you do, you'll find the instructions clear and easy to follow.

I gave the SAC-2K a workout with everything from a desktop PC running Steinberg *Cubase VST* to a full-blown Mac-based Pro Tools/24 Mixplus system. The SAC-2K's integration with Mark of the Unicorn (MOTU) *Digital Performer* is particularly nifty.

TOPOGRAPHICAL NOTIONS

A tour of the SAC-2K must begin with a nod of appreciation to the board's design team for recognizing that aesthetic form and ergonomic function can co-exist beautifully. I'm not often moved to comment on gear's appearance, but the SAC-2K is one sharp-looking control surface. Its contours and color scheme are stylish yet understated, and its layout is spacious while its footprint is reasonably compact.

About half of the SAC-2K's surface is devoted to eight touch-sensitive, motorized 100 mm channel faders and an identical ninth fader for master output level. Each channel fader is accompanied by a Select button, a Mute/Solo button, and a rotary-encoder knob for

controlling pan and a zillion other parameters. All the buttons light up when pressed. The fader section is split down the middle by a double column of ten buttons designed for duties such as switching banks and selecting knob functions.

The faders on the SAC-2K feel smooth and precise, and the motors make updating automation on the fly easy. Once you start mixing with a nice control surface such as the SAC-2K, you realize that mixing with a mouse is like using step-entry to enter notes; you can get the job done, but it's not the same as playing a responsive keyboard controller.

On the first unit I had for review, the moving faders fought me and responded improperly to written automation. A firmware update from Radikal added the capability to adjust the faders' touch sensitivity; a low setting helped a lot. Still, the motors
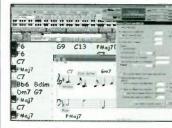
FIG. 1: Radikal's SAC-2K Software Assigned Controller provides tactile control over software-based mixing environments, eight channels at a time. Excellent displays complement the multifunction buttons and transport controls.

were often jittery and imprecise. Radikal reports that static electricity can cause such strange behavior, and though I'm told that many users never have a problem, the ultimate solution involves a factory-installed modification to the faders. Radikal solved the problem by shipping me a second unit with the modification installed. All new units ship with the modified faders, and users who experience problems with older units should contact Radikal to arrange for the free modification.

BANK ON IT

Eight channel faders are more than eight times better than a mouse but still short of the number of tracks many projects contain. Press the bank-select buttons to control additional groups of eight tracks. In *Pro Tools* or *Digital Performer*, you simply press a button to hop from tracks 1 through 8 to tracks 9 through 16 and beyond until you run out of tracks. For programs such as *Cubase VST* that group tracks by category, you first select the type of track (MIDI or audio) and then switch between those tracks with the bank-select buttons. Incremental bank shifting—moving the selection from tracks 1 through 8 to 2 through 9 and so forth—is also possible, letting you control any group of eight adjacent faders.

The rotary encoders are touch sensitive, meaning that they respond with finer resolution as you turn them more slowly, and a quick twist covers a greater range of values. Such natural and efficient behavior beats the pants off pressing a fine-resolution button. Any knob also functions as a button; pressing it selects certain functions or, in some cases, resets a value to its default (such as

center pan or 0 dB). The relative value of the parameter controlled by the rotary encoder is displayed on 31 LED segments that encircle the knob like the spokes of a wheel (see **Fig. 3**).

One of the coolest features is the 6-inch-wide LED text over each group of four faders. Two rows of 40 characters each (10 characters per knob) display the knob parameter, parameter values, track names, and other data, depending on context. Such visual feedback lets you perform many functions without ever looking at your computer monitor. When you use the SAC-2K with *Digital Performer*, the unit even displays the level of your audio tracks using tiny meters.

The remainder of the SAC-2K's surface is devoted to a Jog wheel and additional buttons, including transport controls and buttons for dropping and locating markers. A sizable SMPTE/Locator LED readout is prominently displayed above the Jog wheel. A double row of six buttons above the SMPTE display alternately emulates a computer's numeric keypad and offers one-touch access to edit windows and other functions. At the top of the transport section is another set of four rotary encoders and a third text display like those above the channel faders. Along the right edge is a column of five buttons for controlling the SAC-2K in Channel Strip mode.

Like every aspect of the unit's construction, all of the buttons and knobs feel solid, notwithstanding the button whose lens came off the first time I touched it. (I simply pressed the lens back into place, and it has stayed put ever since.)

FIG. 3: A circle of 31 LEDs surrounds each rotary-encoder knob to indicate its parameter value.

IN THE MODE

The controls have two function sets: Mixer mode and Channel Strip mode. Mixer mode lets you see and control a few aspects of many channels simultaneously, whereas Channel Strip mode lets you see and control many aspects of a single channel. I constantly switched between the modes, using Mixer mode to adjust levels and pan positions and Channel Strip mode to tweak delay times, compressor threshold, and so forth. (Exactly which parameters you can control depends entirely on the host software.) Toggling between the two modes is a snap, and the text displays and lighted buttons reveal the mode you're in at a glance.

In Mixer mode, each channel's rotary encoder functions as a pan, insert, or send control, depending on which button is illuminated in the center bank. When you use the SAC-2K with *Pro Tools*, the Swap function can assign the channel fader to send level and the rotary encoder to send pan, which makes dialing in exact send settings easy. Two buttons just below the knob control mute and solo status as well as track selection. Depending on the host software, you can press the same buttons in conjunction with other button presses to arm tracks to record. All the programs I used made good use of button combinations to maximize functionality.

Changing to Channel Strip mode lets the three text displays show the individual parameters of any plug-in assigned to a track. Selecting a track and pressing the appropriate button in the Channel Strip column puts all 12 knobs at your disposal for tweaking decay time, room size, cutoff frequency, or any other plug-in parameter. If an effect has more than 12 parameters, pressing the same

FIG. 2: On the SAC-2K's back panel, MIDI In, Out, and Thru connectors provide the primary connection to your computer. Once USB drivers are released, the built-in hub will let you connect additional 8-channel fader packs.

Channel Strip button again pages through the additional parameters.

By moving a knob faster or slower, I was able to dial in exactly the settings I wanted with incredible ease. For parameters such as Bypass, I could simply press the appropriate knob to toggle its state. With everything clearly labeled and laid out across the control panel's top, the only time I had to look at the computer monitor was to see the graphic EQ curve or gain-reduction meters on some of the plug-ins.

Because individual programs handle features such as plug-ins, effects, and virtual instruments in many ways, the implementation of the Channel Strip buttons varies quite a bit. *Cubase VST,* for example, has dedicated equalization, which you can access with the SAC-2K's EQ button, for each audio channel. In *Pro Tools,* plug-ins handle the EQ, so pressing the EQ button there opens the selected plug-in's Edit window.

One of the SAC-2K's more imaginative features is its dedicated Instrument button. With any supported software synthesizer, you can use Channel Strip mode to edit individual instrument parameters. The list of supported virtual instruments includes Native Instruments *B4* and *Pro-52,* Emagic *EXS24* and *ES1,* Steinberg *Model E,* and TC Works *Mercury 1.*

SHAPE-SHIFTER

Although the SAC-2K is a solid performer that delivers on the promise of an efficient and professional tactile interface, its implementation varies from program to program. The SAC-2K's completely configurability is its strength, but for some users, it might also be a slight problem. Each software manufacturer is able to customize the SAC-2K's functions to integrate tightly with its host software, resulting in a different set of commands for every program. Learning to use the controller with several programs (as I have) can become confusing. If you use only one sequencing and recording environment and learn the command set well, such confusion will be less of an issue.

The SAC-2K's flexibility begs for comprehensive documentation for each supported program. Online manuals are provided for *Pro Tools* and *Digital Performer,* but the *Cubase VST* documentation is limited to a three-page overview in the SAC-2K manual. The dedicated manuals are quite good, and as a result, getting up and running in those programs was easy. In addition, for those two programs, Radikal has application-specific function templates designed to be placed over the button fields, so you can see at a glance which button controls what function.

Finding my way around *Cubase VST,* however, was mostly trial and error. Radikal Technologies is largely at the mercy of software manufacturers who determine what features to implement and how to operate them within the powerful and flexible framework that Radikal has designed.

Why do control surfaces such as the SAC-2K support some programs so much more extensively than others? Enlightened developers have designed automation-code pathways into their software, enabling their programs to accept various commands from external devices. Some developers, including Digidesign and MOTU, have been doing that for years, whereas others are just now catching up. Consequently, the software developers decide which parameters will offer hands-on control—not

the control-surface manufacturers, who program their firmware to address the automation code.

The implementation of the SAC-2K's controls for *Digital Performer* is exceptional, offering more features than any of the other host programs I tried. One particularly nice touch is that the text displays can show Help information as you work.

Pro Tools support is embedded in the most current firmware (version 1.37 at press time), and you can download a PDF manual from Radikal's Web site. You can also download a *Logic Audio* environment and support files for *Digital Performer* and CreamWare *Scope.* Radikal says that Magix *Samplitude,* Steinberg *Nuendo* and Sek'd *Sequoia* are supported, as well.

Not content to be simply a front end for audio sequencers, the SAC-2K has also been adapted for Emagic *SoundDiver* and Propellerhead *Reason.* Radikal says that *SoundDiver* allows you to assign SAC-2K controls to elements of supported synthesizers by simple drag-and-drop operations.

According to Radikal, support for additional applications is in various stages of development and is forthcoming for Cakewalk *Sonar,* Tascam *GigaStudio,* Sonic Foundry *Acid* and *Vegas,* TC Works *Spark,* BIAS *Deck,* and Ableton *Live.* Also in the works is support for systems from Pyramix, Soundscape, WaveFrame, and E-mu Paris.

SAC-2K Specifications

Faders	(9) touch-sensitive, motorized 100 mm
Knobs	(12) rotary encoders with 31-segment LED rings
Transport Controls	rewind, fast-forward, stop, play, record; (1) Jog wheel
Track Buttons	(9) Track Select; (8) Mute/Solo
Software-Assignable Buttons	(43) organized into three clusters: Mixer mode, Channel Strip mode, software navigation/numeric function
Displays	(3) 2 × 40-character LEDs; SMPTE/locator readout
Ports	(1) USB; (1) 4-port USB hub; MIDI In, Out, Thru; (1) 10 VDC
Dimensions	24.50" (W) × 3.75" (H) × 13.75" (D)
Weight	22 lb. (shipping weight)

Support will likely continue to grow and develop—*Pro Tools* support has already gone through multiple revisions—and keeping up with those changes will require a bit of homework. The reward is enhanced functionality and more efficient means to control your mixes. If the *Pro Tools* and *Digital Performer* manuals are an indication of what's to come for other programs, the SAC-2K's future is bright indeed.

MINOR GLITCHES

The nature of control surfaces dictates that although the SAC-2K might work perfectly well in one environment, its performance may vary in another. When I used *Digital Performer* with a Pro Tools TDM system, for example, the SAC-2K had minor problems with the TDM plugins. Although I could select presets normally, the name of the preset often didn't update properly on the SAC-2K's display. More limiting is the fact that I couldn't directly edit effects parameters with the SAC-2K, as Channel Strip mode apparently doesn't work with TDM effects. (Radikal insists that it should work, but no matter how hard I tried, I had no success.) When I switched from Digidesign Audio Engine (DAE) to MOTU Audio System (MAS) mode, the host-based plug-ins were far more cooperative, updating presets and allowing full parameter editing.

While controlling faders in a group, I noticed that when I moved a fader quickly, the others had a tendency to lag slightly. A glance at the computer screen showed me that the only delay was in the updating of the physical controls; the onscreen faders moved in perfect unison. During playback the physical faders behaved as one.

Probably the greatest restriction in working with the SAC-2K is its limited support for selecting and editing audio and MIDI data. Because support for the SAC-2K depends entirely on the music-software developers, though, the problem is out of Radikal's control. For example, though standard *Pro Tools* commands such as Cut, Capture, and Separate have dedicated buttons, you can select data only by setting markers on the fly during playback; I'd rather

be able to set and adjust markers while scrubbing. Zoom buttons are active only when the Zoomer tool is selected, which decreases their utility. *Digital Performer*'s support for edit functions is better, allowing you to set markers in any transport mode. In general, though, I would like to see software vendors develop better support for edit functions.

FINAL COUNTDOWN

Your assessment of the SAC-2K's value will probably depend on your frame of reference. If you're looking for most of the functionality of a Digidesign Pro Control for a bare fraction of the price, you'll regard the SAC-2K as a tremendous bargain. If you're doing a cost-versus-feature comparison with a CM Automation Motor Mix, however, you may think the SAC-2K is a bit pricey. Anyone looking to upgrade from a basic system such as a Tascam US-428 should do a dollars-to-drool analysis to justify the purchase.

The SAC-2K fits nicely into a midrange position among tactile interfaces. It looks right at home in any studio, without taking up too much space. It's just as comfortable with Macs as it is with PCs, and as soon as its USB issues are resolved, it will be expandable to boot.
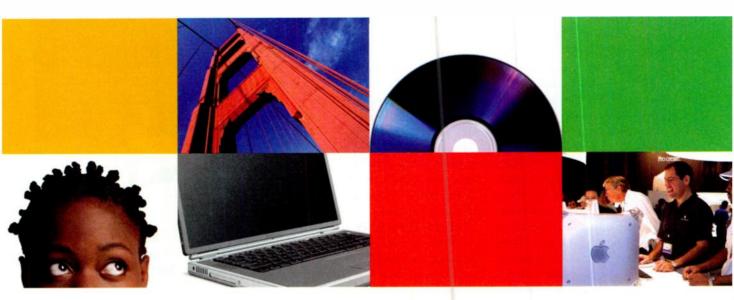
As they would with any control surface, users should evaluate the level of support available for their favorite applications. With the roster of programs already onboard or under development, you'll probably find reasons to like the SAC-2K. In time I hope that all the documentation rises to the level of the *Pro Tools* and *Digital Performer* manuals and that Radikal produces additional application-specific function templates to lay over the buttons.

The more editing I can do from a control surface, the happier I'll be. When I no longer have to touch that rodent next to my computer, I'll probably forget the address to which I'm supposed to return the SAC-2K.

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

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BIG FISH AUDIO

ROOTS OF THE MIDDLE EAST AND NORTH AFRICA AND HADEETH ARABIC RHYTHMS

Stock up on sounds from around the world.

By Jeff Obee

Having played various forms of Middle Eastern music over the years, I am well acquainted with the richness and depth of those powerful and captivating styles. For those who desire authentic Middle Eastern rhythms, whether for world fusion or more traditional uses, Big Fish Audio has two recent productions that warrant a close look.

Roots of the Middle East and North Africa was produced by classical pianist, orchestral arranger, and synthesist Norair Sarkissian and performed by accomplished percussionists Mark Assaf, Gary Jaklian, Fareed Suleiman, and Ghazi Baradah. *Hadeeth Arabic Rhythms* was produced by drummer, producer, and engineer Ara Antranik. Antranik played

Big Fish Audio's *Roots of the Middle East and North Africa* is an extensive collection of hand-percussion loops covering a range of authentic styles.

the drum kit and *sakat* (finger cymbals), and master session musician Haytham Ballat played the other percussion.
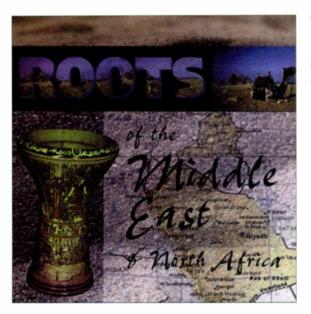
ORIGINS AND INSTRUMENTS

Each collection consists of a single CD. Although their titles are different, both focus on similar ethnic origins and cross into separate styles. The two also cover a broad swath of rhythms from North African countries such as Morocco, Algeria, Tunisia, and Egypt; Saudi Arabia; and the Fertile Crescent (Syria, Jordan, Lebanon, and Iraq).

A number of rhythms appear on both CDs, including *Adani* (from Adan, capital of Yemen), *Ayoub* (Persian Gulf), as well as *Maksum* and *Malfouf* (both popular Egyptian and Fertile Crescent rhythms). You'll also find examples of *Karachi*, *Falahy* (an Egyptian farm rhythm now popular in modern dance music), and *Baladi* (common across the Middle East) patterns. (Go to www.emusician.com/emlinks for audio examples of those and other rhythms from the CDs.) Rhythms unique to the *Roots* collection include *Katakofti, Bambi,* and *Sudasi,* and those only on *Hadeeth* include *Okruk, Aksak,* and *Hachah* (traditional Iraqi). The spellings of those styles differ on each CD.

Roots focuses completely on hand percussion; *Hadeeth* adds drum-kit examples to the hand-percussion loops. Both include one-shots of single instruments at the end of each disc. Several instruments appear on both discs, including the *tabla* (known in the West as the dumbek), the *duhollah* (a large bassy dumbek), and the *tabl* (a two-headed drum that is worn around the neck and played with sticks). *Hadeeth* differs from *Roots* by adding the contemporary tabla (a plastic-head dumbek), finger cymbals, *duff,* and *tar* (a thin-framed drum). *Roots* has its own unique set of instruments, which are the *katem* (similar to a

conga), *mazhar* (a katem with metal rings around it), *merjana, tayaran,* and *taoyan.*

People who are not familiar with music from those regions should be aware that the same instrument can have different names. For example, the *Roots* instrument listing describes the *riq* as a Middle Eastern tambourine and mentions that it is also called a *daff.* However, a daff has a different meaning in some other cultures, in which it refers to a type of frame drum. Moreover, instruments from the Middle East often have similar names. For example, be careful not to confuse a duff (a Persian *zarb*) with the aforementioned daff.

AUDIO QUALITY

Before I learned about the recording methods used, I spent considerable time listening to and using both CDs. *Hadeeth* is decidedly superior in audio quality. The loops have more meat and attack, and single-instrument hits stand out much better when imported into my software sampler (Emagic's *EXS24*) and used in a piece of music.

I also enlisted Middle Eastern percussion virtuoso Kevin Mummie for his expert feedback. After doing a few
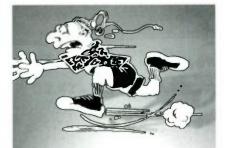
A/B comparisons, Mummie agreed that *Hadeeth* had a punchier sound with more clarity and depth—a quality that *Roots* lacks.

The differences in audio quality can't be attributed solely to the gear:

▼

The discs have lots of varying tempos and some good odd-meter performances.

Roots was recorded with Neumann U 87 and U 47 FET mics, through Avalon and Neve pres and an API console, and then to media, including Soundscape, a Fostex D824, and Tascam 2-inch tape. Editing was done in Pro Tools, Sonic Foundry's *Sound Forge*, and other digital-audio applications using state-of-the-art plug-ins. Antranik wouldn't share what mics and pres he employed, but everything was recorded to a Studer 2-inch analog tape deck. Mastering and editing were done in Steinberg's *Cubase*.

The differences are probably more the result of a pure signal path and one's ears. Antranik not only produced his project but also engineered, mixed, edited, and mastered it. Sarkissian was involved in most steps on his project, but five others were involved as well, perhaps creating a "too many cooks" scenario.

FAVORITE LOOPS
Every track on both discs starts with a full percussion performance that is followed by individual instrument loops. *Hadeeth* contains more individual elements in the main performance and, consequently, has more single-instrument loops in each track. *Roots* has more 16-bar

loops, whereas the maximum length on *Hadeeth* is 8 bars.
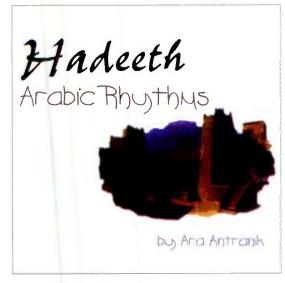
Hadeeth also has far more tracks than *Roots*, weighing in with 40 world-fusion tracks, 10 shorter additional rhythms (no bpm is given for those), 22 traditional tracks, and 11 tracks of one-shots. *Roots* offers 41 full tracks in a mix of styles and 4 tracks of one-shots.

Mummie says both productions provide excellent coverage of styles—as much as anyone would ever need. The discs have lots of varying tempos and some good odd-meter performances.

We particularly liked Aksak on *Hadeeth*, a solid 9/8 rhythm at 120 bpm, and the alluring 11/8 *Samaie Thakil*. The second Falahy loop on the same CD is a joyous 4/4 romp, cooking along at 190. *Hewa*, from Southern Arabia, is a loping 6/8 at 120 bpm with the accent on the fifth beat. It instantly took me back to the hundreds of Persian gigs I've played.

Semmayi Darej from *Roots* is another intriguing odd-meter rhythm in 5/4 at 70 bpm. The Sudasi rhythms are mostly in 6/4 with an irresistible lilt that makes you want to dance. Common rhythms such as Maksoum and *Sayidi* are also well represented on *Roots* with six and eight variations, respectively, at differing tempos.

I also compared loops in the same

Hadeeth
Arabic Rhythms

by Ara Antranik

The well-recorded and -produced *Hadeeth Arabic Rhythms* sample CD contains loops and one-shots. The collection has single-instrument and percussion-ensemble performances.

PRODUCT SUMMARY

Big Fish Audio
Hadeeth Arabic Rhythms
sample CD
$99.95

QUALITY OF SOUNDS	4.5
DOCUMENTATION	2.0
VALUE	4.0

RATING PRODUCTS FROM 1 TO 5

PROS: Includes drum-kit loops. Large assortment of tracks and one-shots. Comprehensive. High-quality sounds.

CONS: Inadequate documentation. Audio CD only.

Manufacturer

Big Fish Audio
tel. (800) 717-3474 or (818) 768-6115
e-mail info@bigfishaudio.com
Web www.bigfishaudio.com

styles from each CD. The common Middle Eastern rhythm Baladi is included in both collections; the *Hadeeth* duff performance from track 27, Baladi No. 2, and the *Roots* daholla performance from track 4 are nearly identical. Each has a distinct character and is beautifully performed, yet *Hadeeth*'s strength shines through. You get the feel of skin on the drumhead and audio clarity throughout the example, whereas the *Roots* loop sounds too midrangey.

DOCUMENTATION AND ISSUES

Both CDs have essentially the same documentation and structure. A list of the instruments with brief definitions of each is provided. The track number is followed by the name of the style, the length in bars, the tempo and meter, and then the names of the individual instrument loops within the track. However, I have some concerns about the material and documentation (or lack thereof).

The collections make good resources for learning Middle Eastern and North African rhythmic styles. But the discs are marketed to a large Western user base, and the context is missing: is the loop urban, folk, classical, cabaret, or peasant village? Those who use the CDs need to be aware that almost all of the rhythms accompany a religious ritual or traditional dance. A short clip with other melodic instruments and an explanation would help users understand the context in which the rhythms are normally played.

Furthermore, users should know that *Middle East* is a generic term. Using Saudi Arabian music in a visual setting in which an Egyptian walks across the street is like playing Cuban music to a scene in which a cowboy walks across the street—it won't make sense. Of all the loops on the two CDs, only one on *Roots* references its country of origin.

In lieu of that, how about including a discography? That would let users at least know the artists who have given these musical forms life over the years and seek them out if they are interested. A certain degree of homage to the original masters is warranted.

In the final analysis, both CDs give quality performances in a variety of styles. *Hadeeth* wins out because of its higher-quality audio; larger quantity of tracks, loops, and single-hit samples; and the addition of drum-kit performances. *Roots* has performances by more players and is also available in CD-ROM format. CD-ROM is easier to use because the files are ready to import into Akai-compatible software and hardware.

Neither disc goes far enough in documenting its source material. *Roots* gives a brief reference on some files (for example, denoting that Baladi is "from the countryside"), and *Hadeeth* distinguishes between the traditional and the modern, but that's as far as it goes.

Whichever disc you choose, you'll get genuine Middle Eastern and North African rhythms performed by skilled musicians experienced in the genres. If your music begs for the real deal, pay the Big Fish site a visit and compare for yourself.

Jeff Obee *is a fretless bassist, synthesist, and composer in the San Francisco Bay Area. Reach him at obeej@dsp.com.*

E M A G I C

SOUNDDIVER 3.04 (MAC/WIN)

An editor-librarian

with a style

all its own.

By Marty Cutler

Since the review of Emagic's *Sound-Diver* 1.5 in the July 1995 issue, the program has undergone several updates. Many new features consist of cosmetic overhauls, changes in nomenclature, bug fixes, and new editors for devices that hit the streets after version 1.5. In addition, several substantive changes deserve closer scrutiny, including the program's beautifully enhanced graphics, support for key commands, and consistency of features between Mac and Windows platforms. *SoundDiver* ships with Mac and Windows versions on the same CD-ROM. Since the release of version 3.0, the two platforms share identical features (except for the differences between Mac and Windows menu conventions and MIDI interface support).

FIG. 1: *SoundDiver's* Install window lets you manually install MIDI devices in your system and provides buttons for automatic scanning and installation.

I installed *SoundDiver* on my Power Mac 8500 running OS 8 without a hitch. The program no longer supports 68000 Macs, but any PowerPC will do. The copy protection scheme consists of a temporary authorization from the CD-ROM. After the hard drive is authorized, the program requests the original CD every few weeks. A week before the authorization expires, *SoundDiver* presents a warning so you won't be caught by surprise. That beats dongles and is better than losing installations from hard-disk crashes, but it's not especially convenient.

WHERE'S MY GEAR?

When first launched, *SoundDiver* scans the MIDI ports to sort out the interfaces and patch bays connecting the MIDI gear and then quickly opens the Install window (see **Fig. 1**). That window provides a list of editors for an array of synthesizers, samplers, drum machines, effects processors, and other devices. At the left of the list are three buttons for installing editors: the Scan All button polls your MIDI system for each device on the list (and takes a while); the Scan button checks only for selected devices. I recommend using the latter. If all of your devices aren't recognized, the Add button lets you select and set up devices manually.

I first tried the Scan All option. After a lengthy assessment of my connections, the Setup window opened to reveal only some of my system's devices (see **Fig. 2**). That isn't necessarily the program's fault. For example, my Casio VZ-10's primitive implementation necessitates manual installation and dumps from the front panel. My Kawai K5m defaults to SysEx-disabled whenever it is powered on.

SoundDiver has added support for Open Music System (OMS), allowing users to switch freely between the program's built-in MIDI system and OMS. I created a new installation using OMS as a MIDI driver; to ensure an accurate setup, I quit

the program and trashed my *SoundDiver* preferences in the System folder. When I restarted the program, *SoundDiver* was initialized and ready to set up. After a new scan of my system, the Setup window showed duplicates of many devices. I continued the installation and deleted the duplicates afterward. When MIDI devices are scanned and accounted for, they appear in *SoundDiver's* Setup window, and a dialog box pops up asking if you want to request the memories from your devices. At that point, you can retrieve the data stored in your devices for editing or safekeeping in a library.

HELLO, INFORMATION

MIDI devices appear in the Setup window as realistic three-dimensional icons, along with a virtual patch bay line connected to a virtual computer. Clicking on an icon opens the Device window (formerly called the Memory Manager). The Device window displays your synthesizer's memory architecture (see **Fig. 3**). I double-clicked on my Korg Wavestation A/D's icon and was greeted with a window containing a layout of the synth's Byzantine memory architecture, including Patches, Wave Sequences, Performances, Multimode setups, Global parameters, and even user scales. That can be a double-edged sword.

With all of that information presented at once, finding your way around (especially in devices as complex as the Wavestation) can be difficult. Fortunately, *SoundDiver* provides several ways to focus on what you need. Buttons at the top of the Device window can hide or reopen any level of the device's memory structure, such as programs, combinations, and Global settings. Furthermore, you can display or hide each bank retrieved from the device, which lets you isolate a single memory area for editing. You can also

zoom the Device window in or out for a broad or narrow overview of the device's contents.

The Library window lets you store anything, from a device's patches to your MIDI system's memory contents. Because libraries can hold just about everything that you need to store, it's important to keep track of related data, and *SoundDiver* nicely handles that task. For example, clicking on a Korg M1 Program in the Librarian window reveals a list of Combinations that use the Program. Similarly, clicking on any Combination provides a list of Programs that comprise the higher-level sound. Deleting a Program opens a dialog box warning you that you're about to remove a component of a higher-level memory structure, thus preventing your carefully sculpted brass-section Combination from sounding like a Martian banjo orchestra or vice versa.

EDITORIAL ASSISTANCE

Double-clicking on a device's memory constituent (such as patch or performance) opens the Edit window. Top-level Combinations that usually provide patch layers, Velocity switching, splits, and so forth have handy edit button links to the lower-level patch components. In that way, you can easily shuttle

FIG. 2: The Setup window displays the contents of your MIDI system with realistic icons. The number to the left of an icon is the device ID.

between the performance and its components, making edits as needed. If you think that seems like too much for one monitor to handle, you're right. However, you can resize windows to focus on what you need and save window settings as Screensets. You can then toggle between screens with key commands.

SoundDiver displays your patch's architecture in gorgeous, color-coded detail. You can change the color of backgrounds, parameters, parameter labels, value fields, and even the handles that shape the envelopes.

In some cases, Emagic's programmers extended features beyond a device's native capabilities. For example, in addition to the typical programming parameters on the Kawai K5000 additive synthesizer series, *SoundDiver*'s Patch editor provides a three-dimensional time-and-amplitude waveform display, a pull-down menu of preset analog synthesizer waveforms, and an assortment of preset Formant Filter settings. Moreover, you can load a digital-audio file for analysis and resynthesis to provide a raw oscillator waveform from which to work. The results provided a good number of interesting timbral springboards for programming. Although that feature was not introduced in version 1.5, the instrument (and consequently, its editor) arrived on the scene after the review in **EM.** Furthermore, the Import function didn't work reliably until version 3.0.

ON THE SURFACE

SoundDiver now supports a variety of MIDI control surfaces, including the Radikal Technologies SAC-2K (see p. 146 for a review of the SAC-2K). I don't own a dedicated control surface, but I installed a

FIG. 3: In this example, *SoundDiver*'s Device window contains the hierarchy of a Korg Wavestation A/D's memory. You can edit all Performance levels simultaneously.

dummy unit in my Setup window to check out Emagic's new drag-and-drop programming feature. As usual, double-clicking on the SAC-2K icon opens the Edit window; however, in place of an abstraction of the device's memory contents, you get an enlarged view of the control surface. To assign controls to a synthesizer parameter, simply open the Control Assignment window, click on one of the virtual SAC-2K's faders or buttons, and drag a virtual patch cord to a synth editor parameter in another window. The intuitive user interface for programming control surfaces let me program the dummy SAC-2K with a knob that could select high or low harmonics in my K5000W synth (see **Fig. 4**). I then assigned a fader to change the gain level of the selected group of harmonics. The entire process took less than ten seconds.

SoundDiver lets you set up a MIDI keyboard as a control surface to edit other synths. You can map practically any incoming MIDI message to control an editing parameter, so even if your master keyboard offers only the standard set of Modulation and Pitch Bend wheels, you can quickly press them into service as editing tools. I opened one of my Oberheim Matrix-6R patches, highlighted the FM Amount parameter, and hit Command + L. I could then adjust my Matrix-6R from my Korg M1 and play while I edited. You can test edits for playability as you make your changes—a very musical approach to programming.

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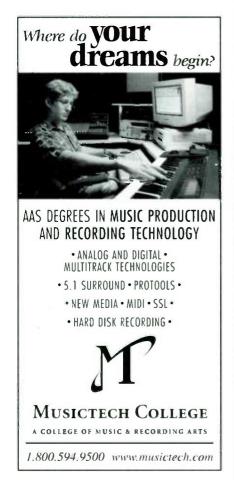
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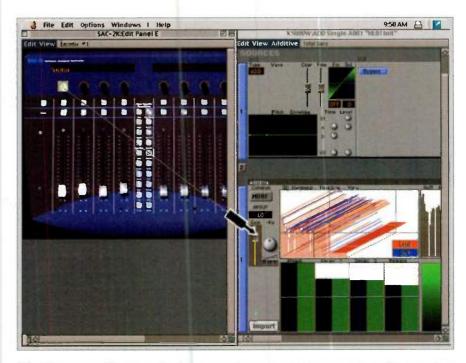
FIG. 4: You can quickly program hardware control surfaces to edit your synths. In this example, a virtual patch cord from the SAC-2K editor's knob has been dragged to control harmonic levels on a Kawai K5000W.

Another clever editing feature is the Overview window. It provides a generalized view of a device's parameters and signal flow, which lets you make coarse adjustments to the patch (see Fig. 5). A single-click just outside a gadget that adjusts parameters opens a full-feature editor with the selected parameter visible in the center of the screen. You can toggle between detail and overview from a field at the top of the Edit window. That's great stuff because it lets you locate and focus on the elements you need to work on, despite a potentially bewildering array of parameters.

Unfortunately, the Overviews are not consistently implemented in all of the editors. The K5000W editor module offers more parameters than could possibly fit on one screen, yet it has no Overview. Thankfully, the editor for the Wavestation A/D does. The work-around is to create multiple Edit windows for a device with each window showing a different parameter area. You can then resize the windows and switch between them. Edits made

in one window are linked to the other windows, but that approach seems like a cluttered and rather clumsy way of focusing on what you need.

You can also grab the bottom-left corner of an Edit window (just before the scroll arrow) and drag directly to any group of parameters. That's a handy navigational tool, but I would most welcome a context-sensitive pull-down menu for jumping to a specific group of parameters (such as the filter section). When navigating a complex
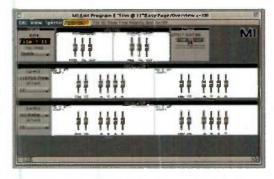
FIG. 5: The Overview window lets you make basic, coarse adjustments to your patch. You can make focused, detailed edits by clicking on any parameter section box. You can toggle back to the Overview with the button at the top left of the window.

device, dragging until you find your area of interest is not the most elegant solution.

ADAPT OR DIE

Adaptations are user-created editors, and the Adaptation Editor does not let you create Overviews, so Adaptations don't have Overview windows. That brings me to my next big gripe: of *SoundDiver*'s 346 editors, 65 are Modules and 295 are Adaptations. It may seem trivial in the face of so many supported devices to carp about the ratio of Emagic-authored to user-created editors. However, Adaptations differ from Modules in many ways, and sometimes their feature sets are minimal. For instance, the K5R Adaptation offers no editors whatsoever; it offers only librarian capabilities. Modules are generally full-featured editors. It's great that *SoundDiver* includes facilities for creating editors; as an end-user, however, I would much rather spend my time with music and sound design than with tables of SysEx values.

The manual is quite good and thoroughly tackles the enormous number of features. Additionally, the CD-ROM contains PDF files with documentation listing supported devices and editable parameters. You also get a programming guide for creating Adaptations and an application for compiling Help files. The software provides extensive context-sensitive help, though the program often quit unexpectedly in the midst of selecting topics.

DIVER DOWN

An editor-librarian program can be far more than an efficient tool for retrieving, managing, and creating patches for MIDI gear. A consistent graphical user interface (GUI) can offer windows into the inner workings of synthesizers by presenting a better visual understanding of the device's architecture and signal flow.

I have strong but conflicting feelings about Emagic's *SoundDiver*. Its powerful, feature-laden graphical editors are often hampered by the sheer amount of information they contain. Fully developed editors run alongside sometimes less-than-utilitarian Adaptations.

The Overview window helps alleviate the information overload, but it's not implemented consistently. However, the software supports a slew of MIDI devices—far more than I'll likely ever own—and some of *SoundDiver*'s features are absolutely inspired. What's more, the editors often provide additional tools that cater to a device's unique features.

SoundDiver's focus on an almost totally graphical user interface provides incredibly intuitive programming. Yet the sheer task of navigating the windows can be tedious and counterintuitive—the interface giveth; the interface taketh away.

Regardless, *SoundDiver* is a powerful and versatile tool, and I look forward to the program's next release. You may not want to wait that long to check it out.

Synth programmer and **EM** *assistant editor* **Marty Cutler** *eagerly awaits an editor-librarian for his VCR.*

SONIC IMPLANTS
String Boxes (GigaSampler)
By Zack Price

....

Sonic Implants' *String Boxes: Mellotron and ARP String Ensemble* CD-ROM library ($69.95) brings together string sounds from two classic keyboards: the Mellotron and the ARP Solina String Ensemble. The Mellotron produced its sounds by playing analog tapes of each note of a recorded instrument, and the ARP Solina synthesized its string and brass sounds from sawtooth waves. Although the Mellotron has always been considered a unique, classic instrument, the Solina has only recently earned its standing as a vintage synth.

Let Me Take You Down
As you might expect from the price tag, this is not an extensive collection of instrument sounds. The Mellotron library is limited to Flute, 8-Voice Choir, and three String sounds from the M200, M300, and Mark II models. However, those are the instruments most closely associated with the Mellotron sound, and they are good representative choices. Furthermore, each sound has a number of variations created

Sonic Implants' *String Boxes: Mellotron and ARP String Ensemble* gathers the classic sounds of the Mellotron and the ARP Solina String Ensemble for *GigaStudio* samplers.

by adjusting envelopes or filters. For example, the M200 Strings library has four filtered pad sounds with different attack and release envelopes, a basic Dry Violins patch, and other pad variations that alter sample playback by using only attack and release envelopes. The Mellotron samples were recorded at a 22.05 kHz sampling rate with 16-bit resolution, which is enough to ensure faithful reproduction of the Mellotron instruments.

I was shocked to discover that each Mellotron sample loops back to the beginning with a noticeable audio gap between the end and the beginning of the loop. When I mentioned that to Sonic Implants, I was told that it was intentional. Fortunately, you can defeat the loops so that the samples will play only from beginning to end like the tapes in real Mellotrons.

String-Driven Thing
The Solina sounds are grouped into five libraries: ARP Ensemble 1 Full Octave, ARP Ensemble 2 Full Unison, ARP Ensemble 3 Strings Only, ARP String Ensemble Mod, and ARP String Ensemble No Mod. The Full Octave and Full Unison libraries are samples of string and brass sounds played an octave apart and in unison respectively. Like the Mellotron libraries, the Full Octave, Full Unison, and Strings Only libraries contain variations based on envelope and filter adjustments. The String Ensemble Mod and String Ensemble No Mod libraries differ from each other in that the Mod library patches have a thicker, more choruslike sound than the sounds in the No Mod library. Both libraries include variations such as Viola, Violin, Trumpet, Horn, and Strings, as well as layered patches of the individual instruments in various combinations. Additionally, there are variations based on different envelope and filter settings, as there are in the other ARP and Mellotron libraries.

The Mellotron and Solina String Ensemble libraries respond to MIDI Program Change, Velocity, Modulation, and Sustain information. I would appreciate a few patches that contain MIDI Control Change messages for altering filter settings. That would have been especially useful in the Solina libraries. Nonetheless, you get more than enough variations to keep you satisfied for a long time.

The Best-Laid Plans
The concept behind this CD-ROM is great: include the best-loved and most used Mellotron sounds with a variety of sounds from a classic analog string box and sell it for a nice low price. The only issue is that the Mellotron sounds contain the sound gap that I described earlier. I hope that Sonic Implants will correct the problem in the next release; in the meantime, it's easy to fix. On the other hand, the Solina String sounds are reproduced well and offer a large selection of textures from which to choose. Having real-time MIDI control over the filter for those sounds would have been an extra treat, but I can live without them for the time being. I can always create those patches from the material already provided.

The *String Boxes* library is far from perfect; however, its comparatively low price outweighs its imperfections, and its blemishes are easily correctable. Besides, there are plenty of raw samples and simple instrument designs that you can use for creating some variations of your own. If you are willing to do a minuscule amount of work to get the collection playing properly, then *String Boxes* may prove to be a good deal for you.

Overall EM Rating (1 through 5): 3
Sonic Implants Network; tel. (781) 641-0063;
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SOUNDMAN
OKM II K Classic Studio
By Karen Stackpole

....

If you've ever wanted to be a dummy head for three-dimensional recording, here's your big chance. The OKM II K Classic Studio Binaural Stereo Headset microphone ($325), from German manufacturer Soundman, is one model in a series of in-ear mic pairs designed primarily for binaural recording. The headset is simple and inconspicuous and looks a lot like ear-bud stereo headphones. The mics are intended for use with compact recording devices that have ¼-inch stereo input jacks, such as MiniDisc recorders and DATman-style devices.

Package Deal

The OKM II K Classic Studio mics are a computer-matched pair of tiny omnidirectional, electret-condenser elements. The capsules are embedded in molded-plastic ear buds that look like miniature yo-yos with six prongs jutting outward on each half. Although they are visually reminiscent of torture devices from the Spanish Inquisition (who expected that?), the prongs accommodate and effectively secure a set of foam screens to protect the capsules and your ears. The headset comes nestled in a black plastic case that incorporates a spooling device for the slim mic cable to safeguard against tangling and damage. An extra set of foam screens is included.

The headset includes an A3 adapter, which supplies DC power to the electret elements. The A3 purportedly offers increased dynamic range and quieter operation over directly plugging the mics in to the ⅛-inch mic jack on a portable recording device. The adapter is housed in black plastic and runs on a 6V battery. The battery is supposed to last 100 hours, and a battery check button activates an LED indicator to let you know if power is low. Features include a switchable –20 dB attenuation pad and a selectable low-cut filter.

The OKM II K headset and accessories come in a black wooden box and fit snugly into a formed plastic tray coated with a red velvety covering. The lid is lined with foam and is also dressed in red velvet, which gives the package a classy look. It also includes an owner's manual and a demo CD offering an introduction to various uses.

Testing 1, 2

I took the mics to a number of live recording gigs and conducted a few controlled tests. I don't own a MiniDisc recorder, so I made a custom cable to use the mics with a portable Sony TCD-10 Pro DAT machine and a Panasonic SV-3800 DAT machine.

For the live tests, I recorded several bands in small- to medium-size venues, with and without sound reinforcement, all with a variety of instrumentation. In addition to playing the dummy-head role, I recommend placing the mics ¼ inch apart on a flat surface to serve as boundary-layer microphones.

For the controlled tests, I binaurally

recorded a drum set (from the player's perspective) and compared that with my usual technique of using overhead cardioid condensers in an XY configuration. I also recorded acoustic guitars with the OKM II K in a boundary-layer arrangement and checked out the A3 adapter's –20 dB pad and low-cut filter. Because the setup was so portable, I couldn't resist using the mics to record ambient sounds and effects while walking around too.

Listen to This

The OKM II K reproduced the original listening experience fairly well, especially when listening back on headphones. Any time I moved my head to check levels, however, the stereo image shifted. When you wear the mics while recording, think "mannequin." The microphones captured ambient sounds quite well with minimum hassle. Although the recordings weren't on par with audiophile levels, they were quite good.

The OKM II K mics don't sound very transparent, but they have a punchy low end with good presence. The highs are a bit gritty and biting, and the upper mids sound slightly distorted. Midrange frequencies between 250 Hz and 500 Hz are lacking; although that characteristic reduces warmth, it makes some rooms sound less boxy. On sources such as baritone sax and upright bass, the mics sound a little thin. Compared with the Oktava MC012s as drum overheads, the binaural mics represent a more dramatic spatial image, but transients are less clear and the sound isn't as full. Nonetheless, the recorded results sound realistic enough.

The microphones are small and low profile. After years of toting around a

Designed for recording with portable devices, the versatile Soundman OKM II K is suitable for use as a binaural headset mic, a boundary-layer mic, or a stereo clip-on mic.

loaded rack, mic stands, and a case full of microphones and accessories for stereo recordings, using the OKM II K gave me an immediate sense of emancipation. The Soundman OKM II K Classic Studio Binaural Stereo Headset microphone is an excellent low-profile option to make club recordings and to collect sounds in the field, but it's not exactly professional quality. At the price, however, the mics make an affordable and handy addition to anyone's low-pressure recording arsenal. Even when you consider its limitations, the OKM II K headset is a bargain for convenience and decent sound quality. If you need to be stealthy while recording, the mics are a fine choice.

Overall EM Rating (1 through 5): 4
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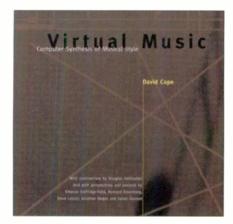
Virtual Music: Computer Synthesis of Musical Style

By Douglas Geers

• •

Since the 1980s, composer and researcher David Cope has taught computers the art of music composition using his software *Experiments in Musical Intelligence (EMI,* or *Emmy).* Over the years, *Emmy* has evolved, writing pieces in the styles of composers such as Bach, Beethoven, Joplin, and even Cope himself. *Virtual Music: Computer Synthesis of Musical Style* ($45) is a collection of writings about *Emmy,* including descriptions of how it works and reactions, musical and philosophical, to the music *Emmy* composes and the idea of the computer as composer. An accompanying audio CD includes all of the book's musical examples, including several pieces written by *Emmy.*

Most writings in *Virtual Music* are taken from a colloquium on *Emmy* that occurred at Stanford University in late 1997, with other materials added later by Cope. Musicians and cognitive scientists who contributed to the book include Douglas Hofstadter, author of the Pulitzer Prize–winning book *Gödel, Escher, Bach: An Eternal Golden Braid. Virtual Music's* multiple-author format provides a pleasing variety of perspectives on *Emmy.*

Virtual Music by David Cope is an exploration of Cope's Experiments in Musical Intelligence (Emmy) software. Emmy "composes" music in various styles based on its analysis of musical examples.

Whither Computer Music?

The book is divided into four major sections: Fundamentals, Processes and Output, Commentary, and Response and Perspectives. The text begins by defining the concept of virtual music as "machine-created composition, which attempts to replicate the style but not the actual notes of existing music," and traces its history from figured bass and musical dice games of the 17th and 18th centuries to today. That provides an interesting background for Cope's work.

Hofstadter offers his thoughts about Emmy in chapter 2. He feels amazed and even a bit fearful of Emmy because it seems capable of writing music that can touch his emotions just as strongly as music written by humans. He makes some of his argument in poetry: "Can one bypass the soul, / Can one sidestep all strife, / And produce wondrous music, / Without living a life?" Hofstadter's comments and novice's explanations of how Emmy functions are well stated and provide an intriguing counterpoint to Cope's own descriptions.

Cope follows in chapters 3 through 6 with a response to Hofstadter and an overview of how Emmy composes. Put simply, Emmy creates new music by sewing together bits from existing music that Cope inputs. Rather than patching together random chunks of music, Emmy recognizes, remembers, and recombines stylistic elements from its models and then creates new pieces from them. Cope focuses on the concepts that underlie Emmy and avoids long sections of programming code.

In chapters 7 through 10, Cope traces the complete process of using Emmy to create a new piano-sonata movement in the style of Mozart. He provides the reader with two contrasting movements composed by Emmy during that project: one that he labels a success and one that is a reject. The score and a recording of each are available for easy comparison.

Although that section clearly lays out Cope's methods, it dives into quite a bit of detail, possibly too much for many readers. Listening to the two Emmy compositions proves the value of Cope's research, though. Both compositions work as music, and if someone played the successful piece, I could really believe that Mozart composed it.

The next section, Commentary, examines Emmy's relation to music history, how it succeeds and fails in its imitation of master composers, how people should listen to Emmy's compositions, and how the music might be regarded in the future. Cope then ends the book in chapters 17 and 18 with a look into the future. Some predictions sound a bit too grandiose, but because his current work seems impossible to many people, I respect his opinions.

Emmy Awards

Virtual Music is a joy to read. It is written in a friendly style with the layperson in mind. Although knowledge of classical music and the ability to read music notation certainly help, neither is necessary to enjoy the story and discussions of Emmy within it. However, the text gives the reader no specific descriptions of how he or she might follow in Cope's footsteps and write similar composing software. This is a book to read for its intriguing ideas, not for learning practical skills, and thus it is clearly not for everyone. Lastly, though having a CD of Emmy's music with the book is wonderful, it would have been much more satisfying to hear live musicians play them rather than Cope's Performance algorithm and a sampler. ☻

Overall EM Rating (1 through 5): 3.5

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It All Comes 'Round Again

*"Meet on the Ledge,
We're gonna
Meet on the Ledge,
When you really mean it,
It all comes 'round again."*
—Richard Thompson, "Meet on the Ledge"

The following story is completely true, and I play the proof. Although I'd been banging on things for years, it wasn't until I was 13 years old that I actually began to study drums. My first drum was a Slingerland snare I bought for $5 from my cousin who found he wasn't as interested as he had thought.

I learned on that drum, and in no more than a year, I was ready to move to a full drum set. My starter kit was a cheap four-piece job that arrived from Japan bearing a fabulous blue sparkle finish and the brand name Star. I dug that blue sparkle so much that I traded in the Slingerland so that I could get the matching snare drum.

I leaned on that kit, and in no more than a year, it was starting to fall to pieces, which bothered me doubly because, by that time, I had come into the knowledge that a Slingerland was one of the best drums around. I guess I wasn't supposed to keep it, because I didn't really understand what to do with it.

Flash forward four years to Albany, New York, where I lived as a music major at the State University of New York. In the intervening years, I had played the blue sparkle kit nearly to splinters and saved enough money to buy a "Ringo" kit: a five-piece Ludwig Hollywoods with a black pearl finish. By the time I was living in Albany, I'd been playing them for several years and had become a pretty good drummer.

It was the early part of a lovely fall evening as I walked out of my apartment in Albany. It was trash night, and there was . . . *stuff* on the sidewalk. Reaching the curb, I saw that someone was tossing out the generally smashed-up carcasses of a cheesy drum set.

Then I noticed that pretty much the only thing that the snare drum had in common with the rest of the set was a wood finish. Curious, I picked up the snare and looked it over. As I rotated it, the nameplate came into the streetlight: Slingerland. I'm not making this up; I still have this drum, and I'd even bet the streetlight is still there, too, in spite of my nearly knocking it over with a 24-foot truck when I moved from Albany in the middle of a blizzard. Even weirder is that it is actually an older model than my Ludwigs and has six lugs instead of the ten that are more common now.

But wait! There's more.

Throughout many years of playing, I have accumulated drums here and there, mostly from friends who weren't playing them. I got a fabulous kick drum and a tom that match my Ludwigs, a semitrashed snare, and an old Rogers marching drum that I turned into a floor tom. At one point or another, I have used each of those except for the snare because it needed some work, and I already had the Ludwig and the returned Slingerland snares. In fact, it wasn't until a few years ago that I took a serious look at what needed to be done to fix that snare.

I had never given a second thought to the snare's blue sparkle finish until just a few months ago when I happened to spot the nameplate, which said—you guessed it—Star. You just can't make things up as far-fetched as what real life sometimes hands you.

That little affair has actually had a pretty significant impact on my view of life. Clearly, the drum I couldn't appreciate came back to me when I was ready for it, and the Star snare came back to close the circle and make the point that, as the song says, "When you really mean it, / It all comes 'round again."

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.*

STEVE CURL