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Electronic Musician®

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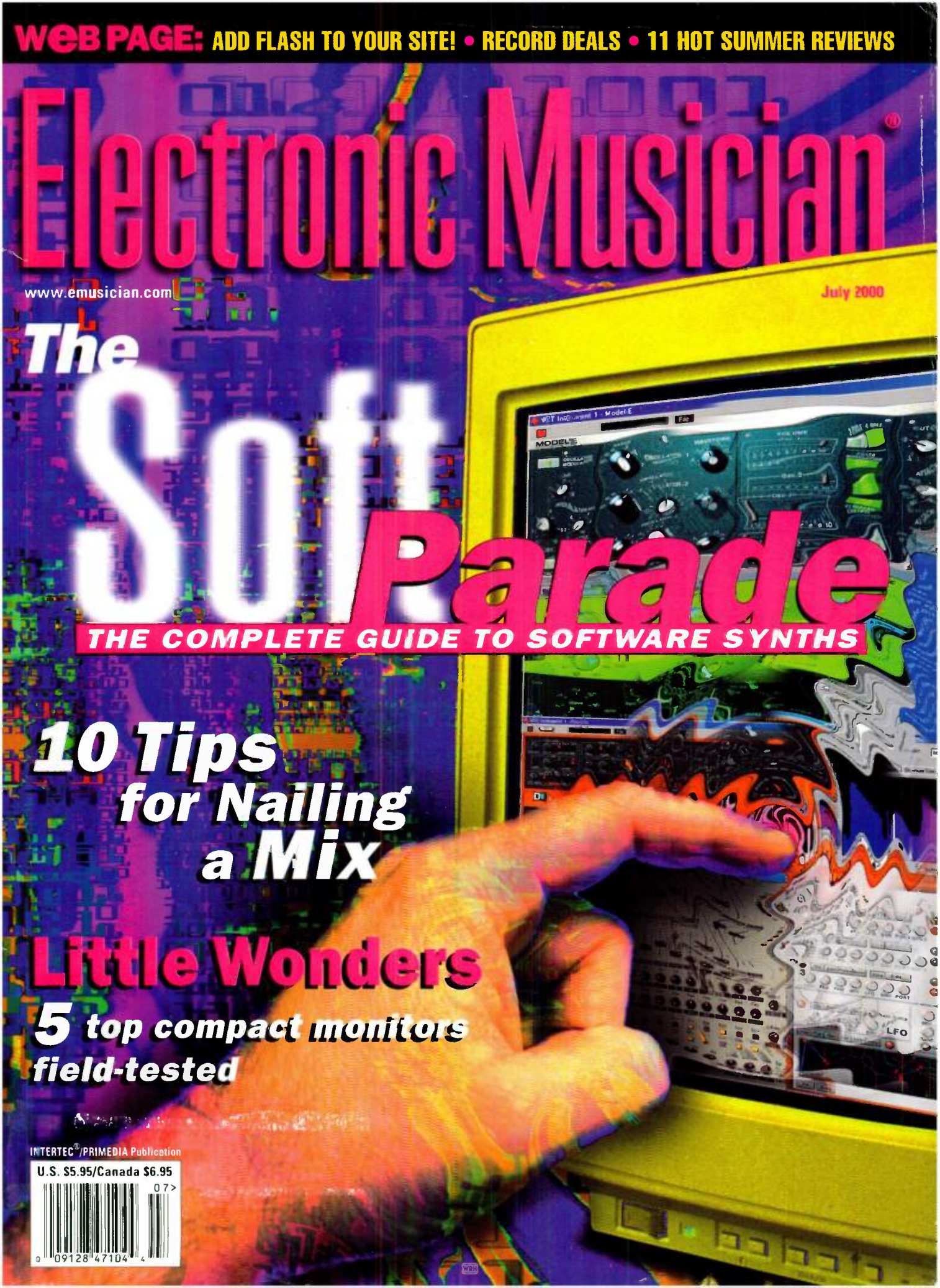
July 2000



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And it's also a full-fe

disk recorder works over 90 minutes

blazing-fast internal processor speed for fast multitrack punch-ins.

Easy back-up.

Equally as important, the HDR24/96 is the first hard disk recorder to satisfactorily address the problem of transfer and storage. Its built-in Ethernet "*HDR Bridge*" lets you FTP data from the recorder to your computer's desktop, existing SCSI peripherals and the outside world at the near-realtime speed of 100Mb/second. The reasonable cost of our pull-out M•90 media cartridges means you can afford to have one for each project.

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FEATURES

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DEPARTMENTS

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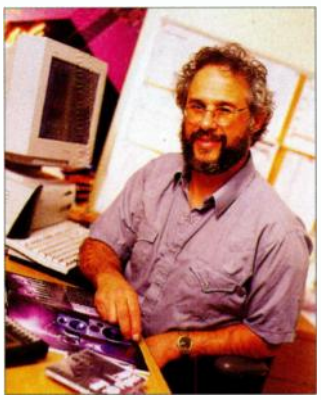
JULY 2000 VOL. 16, NO. 7

COLUMNS

REVIEWS

Simple Pleasures



ANTHONY PIDGEON

It's easy to get lost in today's technological marvels. My studio/home office is a cornucopia of hardware, the heart of which is my Power Mac, simultaneously one of my favorite tools and the bane of my existence.

I was hot to buy a new G4 with Mac OS 9, but the thrill is gone—at least, for the moment. Switching to the G4 and OS 9 has been far tougher than migrating to previous Mac operating systems and hardware because so many technologies changed at once. Perhaps the hardware problems would have occurred with previous models, but OS 9 has been a tragedy for me. After I finally worked out most of the early problems, Apple's OS 9.0.4 updater screwed up my system. (I'll bet I'm the only one who had that experience!) I have raged at OS 9's incompatibility with OMS, suffered with orphaned and obsolete printer and scanner drivers, and a lot more. After this experience, Mac OS X is one Pandora's box I am not ready to open. I have had all the cutting-edge computing I want for a while, thank you very much.

Many people had far smoother migrations to the new Macs, but I doubt my experience is unique. Apple has pushed users to adopt technologies that are not quite ready and in some cases are not even properly supported in the Mac OS. To make things worse, Apple has not yet addressed musicians' specialized needs. For example, it could start by cleaning up the OMS/MIDI mess and ensuring that one can stream audio reliably to FireWire and USB drives from any audio program. Instead, Apple seems to be backing away from its recent initiatives for improving the Mac's music-production capabilities. If Apple truly wants the Mac to remain a popular platform for multimedia-content providers, it must make good things happen for musicians and music-product developers.

Apple isn't the only computer giant whose music support has been spotty. Microsoft has stumbled recently with Windows 2000 and Millennium. Ah, but that's a story for another day.

How can we musicians respond? Obviously, we should look carefully before we leap into bleeding-edge technologies. But then, we electronic musicians like to experiment, and our software tools often require a lot of computing power. Aside from upgrading more cautiously, all we can do is set aside plenty of troubleshooting time.

We can also be less dependent on computers. I have temporarily returned to more traditional electronic music making, using a minimal system comprising an 88-key keyboard synth and headphones, to which I add a mic and a mixer when I want to sing. For tracking, I use an MDM, not the Mac. For effects and more synth and sampler voices, I turn to tried-and-true outboard hardware. The minimal version of this setup, in particular—a keyboard and cans—is reliable and fun to use. Best of all, it lets me focus on music instead of gear.

Make no mistake, I will still enjoy making music with the Mac when it works, because it allows me to do things I could not easily do otherwise. In truth, I am still fascinated with the darned thing. But while my computer continues to cause headaches, I am rediscovering the simple pleasure of just playing music.

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

Subscriptions, Customer Service
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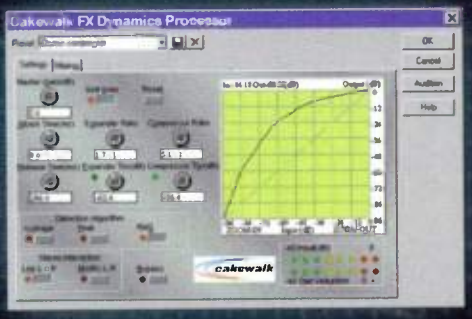

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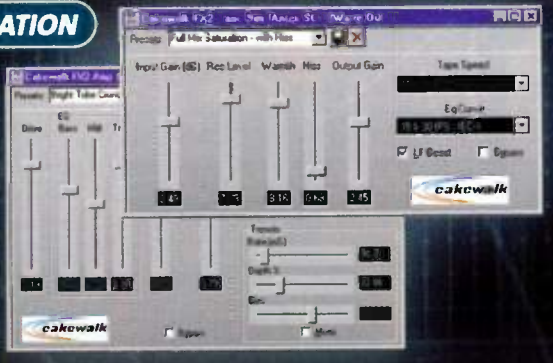

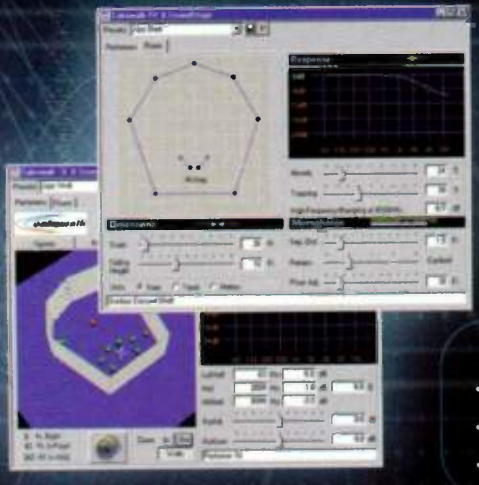

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Hasn't anyone in the audio industry caught on to this?

Joe Garcia
Tempe, AZ

Joe—Thanks for your letter. In fact, a number of PC digital audio programs can employ dual processors running under NT. For example, along with Vegas Pro, *Steinberg's* Wavelab *3.0 and SEK'D's* Samplitude *are multithreaded and will address both processors. Even Cakewalk's* Pro Audio *can use dual processors to accelerate audio-processing tasks.*

I have to warn you, however, that I've experienced many compatibility problems with the dual-processor, dual-boot systems I've had over the past ten years. Several of my audio cards wouldn't work in a dual-processor system, and recently I learned of a software application whose install routine won't work in this system. So check with the manufacturers of the products you're considering to be sure that they're dual-processor compatible. You'll be amazed to learn how many companies have never even tested their products on such a system.—Dennis Miller

DOCTOR'S ORDERS

In the April 2000 issue of **EM**, I read the letter from Mr. Rex Perry regarding the use of *Ginkgo biloba* for the treatment of tinnitus (ringing or noise perceived to be in the ears).

At present, tinnitus has absolutely *no* cure, and there is no scientifically proven relationship between the ingestion of ginkgo biloba and diminished perception of tinnitus.

We see a significant number of patients who have tinnitus in conjunction with hearing loss, and none of the patients who have tried ginkgo biloba as a treatment for the tinnitus have reported any change in the quality, intensity, duration, or frequency of their tinnitus episodes.

Tinnitus is often reported clinically in conjunction with both noise exposure and hearing loss—especially high-frequency sensorineural hearing loss, which often results from noise exposure. Remember that even acoustic instruments such as the violin and piano can create sounds loud enough to damage hearing over time.

To date, ginkgo biloba is unregulated by the Food and Drug Administration. Therefore, adequate dosage information for the average adult user is unavailable. The best treatment for tinnitus is hearing protection and the prevention of hearing loss caused by noise exposure. Interested persons should seek the counsel of an audiologist and otolaryngologist for information. The audiologist should obtain a baseline audiogram in order to monitor changes in hearing acuity over time, and the physician can counsel patients on dietary and lifestyle issues related to tinnitus (for example, sodium, caffeine, tobacco, and alcohol).

Finally, a number of very good musician's earplugs that can be filtered for particular instruments are currently on the market. And the work of Marshall Chasin is an excellent reference for any musician who wishes to protect and preserve his or her hearing.

May is Better Speech and Hearing month, and participation by your publication in raising awareness of the need for hearing protection is warranted.

Sara Ashburn-Reed, M.Ed., CCC/A, F/AAA
The UCSF Douglas Grant Cochlear Implant Center

HELPING HAND

I just have a quick addition to David Rubin's reply to Felix Herzog's letter in the May 2000 issue of **EM**. Everything that David says is true for a *data* CD. Adaptec's *Easy CD Creator* will let you burn an audio CD using MP3 files, but in the background it converts them to Red Book audio format, so there is no gain in the amount of time available, but your regular CD player can play it. I hope that's helpful.

Dave Walker
via e-mail

I NEED SOME SPACE

In my small home studio, I have two Alesis Monitor Ones connected to an Alesis RA-100 amplifier on a shelf a few centimeters above my computer monitor. Each time I turn on the amplifier, the image on the monitor starts shaking. How can I isolate these interferences?

Paulo Jorge da Silva Costa
via e-mail

Paulo—The problems you are experiencing are common in the personal studio. Because desktop space is at a premium, the various components in a recording system are pushed close together in order to maximize this space. Unfortunately, some of the components, such as your computer monitor, are very sensitive to electromagnetic interference (EMI)—those dreaded emissions from the large magnets within monitor speakers. It is likely that the EMI from your unshielded Alesis monitor speakers is making the image on your computer VDT screen wobble.

The problem, however, is easy to fix. Simply move your speakers away from your computer screen. A distance of 8 inches will usually do the trick. While you're at it, put some distance between your power amp and the computer screen as well.

If you simply must have your monitors next to your computer screen, consider investing in a pair of shielded monitors. These monitors have a shielded enclosure that prevents EMI from escaping and causing problems with your equipment. For more information on shielded monitors, see the article "Little Wonders" in this issue of **EM**.*—Gino Robair*

FOND FAREWELL

It is with great sadness that I write to inform you that electronic musician David Michael Matuch (featured in "Creative Space: Home Electronics" in the May 1997 issue of **EM**) passed away on April 14, 2000, at the age of 35 from complications due to his lifelong battle with muscular dystrophy. David completed his third electronic/ambient CD, *Brainstorm: Adventures in Hypoxia*, shortly before his untimely death.

Interested readers can purchase *Brainstorm* as well as David's two previous releases by contacting me at JD990@aol.com. All proceeds will go toward the family's funeral and medical costs.

Michael Mooney
via e-mail

ERROR LOG

May 2000, "What's New," p. 28: The faders on the Roland VS-1880 portable digital studio are not motorized.

March 2000, "To Tell The Truth," p. 62: Guitar Center regularly sells the Oktava MC012 for $249.

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

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WHAT'S NEW

By Marty Cutler

SWIVEL SYSTEMS SG20

Owners of Palm computing devices can now create music with a 16-bit, 44.1 kHz General MIDI sound set. The Swivel Systems SG20 ($200) is a General MIDI tone module that clips onto a Palm computer; it's about half the size of the Palm, and runs on two AAA batteries.

The unit offers 24-voice polyphony with 128 musical instruments and 140 drum sounds. Chorus and reverb are built in. With the supplied MIDI/Audio Expander cable, the SG20 can also be used as a sound module with other MIDI devices. A built-in stereo mini headphone jack is provided for private listening. The SG20 is compatible with all Palm models; to use it with the Palm V and Palm Vx, you'll need the Palmdock V ($35) from Solvepoint.

The SG20 comes bundled with miniMusic's *NotePad* and *BeatPad* sequencing software (see "What's New" in the June 2000 issue of **EM**). Swivel Systems; tel. (415) 474-7446; e-mail simon@swivelsystems.com; Web www.swivelsystems.com.

WIARD SYNTHESIZER MODULAR SYSTEM

The Wiard Synthesizer Modular System is a true analog synthesizer that is available in several different configurations. The standard system ($3,295) is a rack-mountable package, and individual modules are sold separately ($399 to $549). The modules can interface easily with other modular systems using patch cords.

There are two types of VCO modules; one type has four waveforms, an AR envelope generator, a voltage-controlled amplifier, a noise source, and a jack multiple. The other, dubbed Waveform City, adds a digital waveshaper, control-voltage quantization, and nonlinear audio processing. The filter module offers a multimode filter with voltage-controlled cutoff, resonance, and filter type. The filter has a 4-input mixer and jack multiple, which is a patch bay–type splitter.

The Sequantizer is an 8-step analog sequencer module with voltage-controlled stage selection, a voltage quantizer/summer/transposer, and a jack multiple. The Envelator module adds two multimode envelope generators. A switch enables either or both of the envelopes to retrigger at the end of their cycles. This causes the envelopes to ramp up and down, effectively configuring them as LFOs with variable sawtooth or triangle waveshapes. The envelope generator also contains a voltage-controlled crossfader that can be used to crossfade control voltages as well as external audio signals.

The stock system offers a dual-joystick controller. Wiard's optional Manual Controller ($299) comes with two joysticks, two triggers, two stereo patch bays, and a voltmeter. The patch bays offer multiple connector types for interfacing with external audio gear and other synthesizers. Wiard Synthesizer Company; tel. (414) 327-6316; fax (414) 546-9785; e-mail sales@wiard.com; Web www.wiard.com.

PRESONUS DIGIMAX

PreSonus's DigiMax ($1,499) is an 8-channel mic preamp that provides a host of digital and analog I/O features and more. Each channel has a dual-servo mic preamp that does not use capacitors, and each features a peak and RMS limiter with knobs for adjusting the threshold and gain. The first two channels offer high-impedance inputs on the front panel for easy access; a button for each of these channels allows you to reverse the polarity. An EQ-enhancement button is useful for adding "air" around 12 kHz and to subtly contour lower frequencies. The 48V phantom power can be defeated individually for each channel.

The DigiMax comes with eight ¼-inch TRS balanced outputs.

Each channel has its own 24-bit ADC, and you can select sampling rates of 48, 44.1, or 32 kHz from the front panel. The DigiMax offers BNC word-clock I/O and an ADAT Lightpipe output. In addition, separate breakout cables provide four stereo coax S/PDIF ports (eight channels) and four stereo XLR AES/EBU outputs. (You can connect either the S/PDIF or the AES/EBU breakout cable to the unit's single 9-pin port.) PreSonus Audio Electronics; tel. (800) 750-0323 or (225) 216-7887; fax (225) 926-8347; e-mail presonus@presonus.com; Web www.presonus.com.

▲ GVOX INTERACTIVE MUSIC

NotationStation.com is a Web site from GVox where teachers can compose, edit, and listen to music. In this way, they are able to create lessons and post them online for their students.

NotationStation is compatible with Standard MIDI Files as well as native files from GVox's *Encore* and *MusicTime* music-notation applications. Music can also be input with a mouse, computer keyboard, or MIDI instrument. Students and teachers can edit, transpose, change time signatures, and audition individual tracks or an entire piece.

The *NotationStation* plug-in is free and can be used with the Netscape *Navigator* 5.0 browser on Windows PCs. (A Macintosh version is due shortly.) Minimum requirements are a Pentium II/233 MHz PC with 32 MB of RAM, Windows 95 or higher, and either a sound card with a game port MIDI connection or a MIDI interface and an external MIDI keyboard. Teachers can download the entry-level *MusicTime* software for free. GVox Interactive Music; tel. (215) 922-0880; fax (215) 922-7230; Web www.gvox.com.

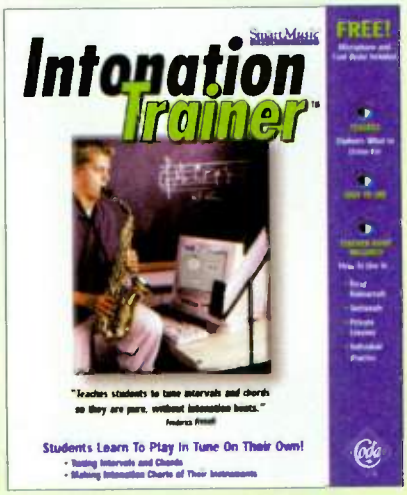
▶ CODA MUSIC TECHNOLOGY

Playing in tune can be one of the most daunting challenges when you're learning a musical instrument. Coda Music Technology has released the Intonation Trainer ($149.95), a software teaching tool that enables students of wind and brass instruments to both hear and see when they are playing in tune. The Intonation Chart Maker feature displays the intonation of every note played; alternate fingerings can also be charted.

The program teaches students to stay in tune when playing intervals or chords in just intonation. You can also learn how to diagnose which notes on your instruments are flat or sharp, thereby anticipating intonation problems.

The Intonation Trainer comes with the software on CD-ROM, a microphone, and a footpedal that allows you to operate the program and still keep your hands on the instrument. A printed teacher's guide shows how to use the package in band rehearsals, sectionals, private lessons, and individual practice.

Minimum requirements for the Macintosh are a Power Mac 603e/180, Mac OS 7.6.1, 32 to 64 MB of RAM, and a serial port for connecting the footpedal. Windows users will need at least a Pentium/166 MHz, Windows 95 or 98, 32 MB of RAM, a Sound Blaster 16–compatible sound card with full-duplex audio, and support for DirectX. Coda Music Technology; tel. (800) 843-2066 or (612) 937-9703; fax (612) 937-9760; e-mail finalesales@codamusic.com; Web www.codamusic.com.



▼ ONLINECONSERVATORY.COM

Despite advances in interactive music-education software, nothing can equal a one-on-one lesson with an experienced music teacher. OnlineConservatory.com has combined the advantages of computers, the Internet, and MIDI with the benefits of personal music instruction. This Internet-based service allows teachers, students, and operators of music schools to work together from any location on the globe.

With a MIDI keyboard, a computer, and *OnlineConservatory* software, interested parties can meet at the OnlineConservatory.com Web site. Teachers can set up their own online school from the site, or join the staff of an existing school. The software application that enables teaching online is designed for the student as well as the teacher: they both have the same display on their screens, simplifying the technological aspects of the session. Student and teacher can talk to each other and play in real time.

The software requires at least a Pentium PC with 16 MB of RAM, Windows 95 or 98, a 28.8 Kbps Internet connection, a MIDI-compatible sound card, and a MIDI keyboard. OnlineConservatory.com; tel. (709) 758-0340; fax (709) 753-0255; e-mail students@onlineconservatory.com; Web www.onlineconservatory.com.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SOUND ADVICE ▲▲▲▲

▲ Q UP ARTS

Q Up Arts' *Dance Partition* (audio CD, $99; Akai CD-ROM, $199) was produced and recorded in Germany and contains 1,500 single-shot modern synthetic drum and percussion samples. Sounds are tailored toward classic dance-floor, techno, hardcore, hip-hop, rap, pop, trance, and jungle music.

Voices of Istanbul (audio CD, $99; Akai, Roland, SampleCell, or GigaSampler CD-ROM, $249) contains single hits and rhythmic multi-instrument loops of authentic Turkish music. The sounds, recorded in Istanbul, feature such instruments as the ramadan drum, ney, tambour kudüm, clarinet, kemence, and darbuka, as well as typical string motifs.

Symphonic Fields Forever (Akai, EOS, Roland, SampleCell II, or GigaSampler CD-ROM, $199) includes multisamples of solo orchestral instruments, trios, and larger symphonic ensembles. This collection is useful in a variety of composing and scoring situations (such as pop or orchestral writing), and the acoustic textures can serve as natural-sounding pads. Multisamples include cellos, violins, silver flute, bassoon, trombone, and choir. You also get multisampled ensemble versions of these instruments.

Drummer Paul Kodish has worked with a number of bands, including the Chemical Brothers, Prodigy, Apollo 440, and Maximum Roach. Q Up's *Kodish Drum N' Bass* (audio CD, $99; Akai, EOS, Roland, SampleCell II, Unity DS-1, or GigaSampler CD-ROM, $199) gathers drum 'n' bass grooves and single hits from electronic and acoustic instruments. Kodish plays an acoustic drum kit while Apollo 440 cohort Cliff Hewit handles electronic hits and subsonic effects. Q Up Arts; tel. (800) 454-4563 or (530) 477-8128; fax (530) 477-5935; e-mail sales@quparts.com; Web www.quparts.com.

VIRTUAL REALITY SOUND

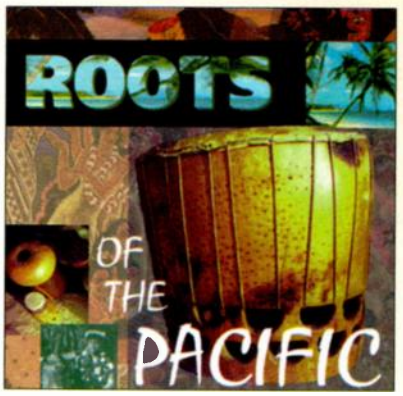
If you'd like to experience virtual reality with your NemeSys *GigaSampler*, *VRSound Giga Module* ($249) may be just the ticket. Virtual Reality Sound's first CD-ROM for *GigaSampler* contains samples of drums, percussion, pianos, bass, synthesizers, and more—all recorded in binaural stereo, using head-related transfer functions (HRTFs).

Also available from Virtual Reality Sound is *Brad Dutz VRSound Percussion* (Akai or Yamaha CD-ROM, $149). Virtual Reality Sound; tel. (714) 240-9175; fax (714) 240-8974; e-mail info@vrsound.com; Web www.vrsound.com.

▼ KID NEPRO PRODUCTIONS

The King's AIFF & WAV File Collection CD ($250) from Kid Nepro Productions boasts an assortment of samples ranging from hip-hop drums to orchestral strings and brass hits, and from funk guitars to classic synthesizer sounds. Synth samples include sounds from the Minimoog, Prophet-5, Wavestation, Voyetra-8, and others.

Sounds are arranged in groups of 100 files, and each file contains multiple samples. Individual files from this collection are available for $5 each and can be downloaded from Kid Nepro's Web site within 24 hours of purchase. Kid Nepro; tel. (718) 642-7802; fax (718) 642-8385; e-mail kidnepro@aol.com; Web www.kidnepro.com.



▲ BIG FISH AUDIO

Big Fish's *Roots of the Pacific* (audio CD, $99.95; Akai S-1000 CD-ROM with audio CD, $199.95) focuses on sounds and instruments of the Pacific Islands. Samples include percussion construction kits, individual instrument hits and patterns, vocal chants, spoken-word selections, and ambient sounds.

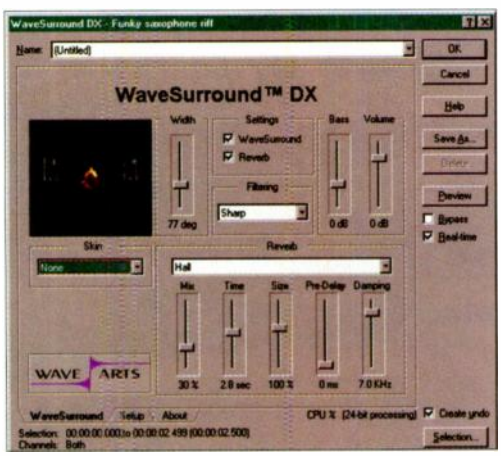
The CD features ensemble and solo drum performances from the Hawaiian Islands, Fiji, Tahiti, and other locales. Percussion instruments include the *pahu*, which is the equivalent of a bass drum; the *to'ere*, a type of slit or log drum; the *ipu*, or bottle gourd; and many more. Chants and samples of *'olelo* (Hawaiian words), along with location recordings of bamboo-forest sounds and other natural ambiences, add to the authenticity of the collection. Big Fish Audio; tel. (800) 717-FISH or (818) 768-6115; fax (818) 768-4117; e-mail info@bigfishaudio.com; Web www.bigfishaudio.com.

SYNERSONICS RESEARCH CONDUCTOOL

If you're plagued with noisy and unreliable cable connections, look into Synersonics Research's ConducTool ($29.95). Using a new patented, oil-free chemical process, the ConducTool is a handy gadget that improves the conductivity of ¼-inch jacks and plugs by cleaning and lubricating them in one fell swoop. One side of the ConducTool has a plug for cleaning ¼-inch jacks; the other end has a jack for cleaning ¼-inch plugs. The tool eliminates the need to disassemble gear such as electric guitars, mixers, and effects boxes for cleaning.

The ConducTool uses a carbon-graphite technology to scrub and lubricate connections without abrasion, leaving an extremely light coating of conductive graphite. Because of the graphite lubrication, the ConducTool will not damage gold-plated connections. The graphite also forms a barrier against oxidation that leaves no deposit as it wears off; after 90 days, it turns into harmless carbon dioxide gas instead of rust or sound-degrading metal oxides. Synersonics Research; tel. (619) 280-2326; e-mail synersonix@aol.com; Web www.conductool.com.

WAVE ARTS WAVESURROUND DX



Wave Arts has released a DirectX-compatible reverb and virtual surround/stereo enhancement plug-in. *WaveSurround DX* ($89.95) features a high-quality reverb that includes five reverb models—chamber, club, hall, stadium, and room. You can change the settings for speaker placement, bass, and volume. The sound can be placed anywhere around the listener and provide a wide or narrow stereo image. The reverb's mix, time, size, predelay, and damping parameters are also adjustable. The plug-in can be optimized for speakers and headphone monitoring as well.

WaveSurround DX works with Cakewalk *Pro Audio,* Syntrillium *Cool Edit Pro,* Emagic *Logic Audio,* Sonic Foundry *Sound Forge,* and other DirectX-compatible programs. Minimum system requirements are a Pentium II/266 MHz computer running Windows 95, 98, NT, or 2000. Wave Arts; tel. (781) 646-3794; fax (781) 646-7190; e-mail info@wavearts.com; Web www.wavearts.com.

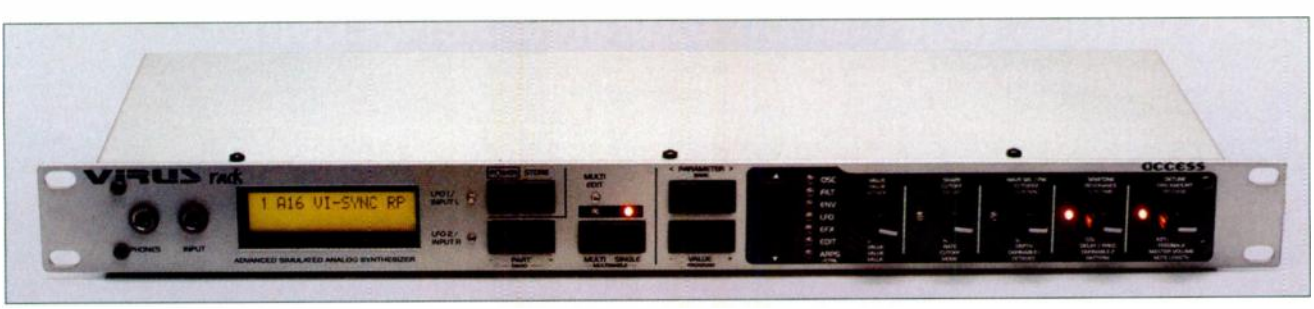
ACCESS VIRUS RACK

The Virus Rack ($995) from Access is a rack-mount version of the company's Virus analog-modeling keyboard synthesizer. A ¼-inch mono input on the front of the Virus Rack allows line-level external audio to be processed through the entire synthesizer architecture. There is also a stereo input on the rear of the unit, and both mono and stereo inputs can be used in a send/return loop.

The synthesizer architecture features three oscillators per voice with a total of 65 waveforms, including variable-pulse, sine, and triangle waves. Each voice has its own multimode filter, which can be configured as lowpass, highpass, bandpass, or band-reject. You can assign saturation and distortion effects for each voice. Three LFOs are available, and the two envelopes per voice offer ADSTR stages. (The T stage is a sustain-time parameter.)

A modulation matrix with more than 100 destinations provides MIDI control. An Adaptive Control Smoothing feature allows you to create subtle and continuous parameter changes. The LFOs, delay effects, and each of the unit's 16 independent arpeggiators can be synchronized to MIDI Clock or to an internal master clock. You can alter the arpeggiator parameters in real time. Effects include reverbs, delays, distortion, vocoder, and Groove Delay (a digital delay with rhythmic patterns).

Analog-to-digital conversion is 18-bit, and the four outputs offer 24-bit digital-to-analog conversion. The Virus Rack comes bundled with a stand-alone copy of Emagic's *SoundDiver* editor-librarian software. Access/GSF Agency/TSI International Sales (distributor); tel. (310) 452-6216; fax (310) 452-3886; e-mail gsfa@netcom.com; Web www.access-music.de.



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▲ STEINBERG

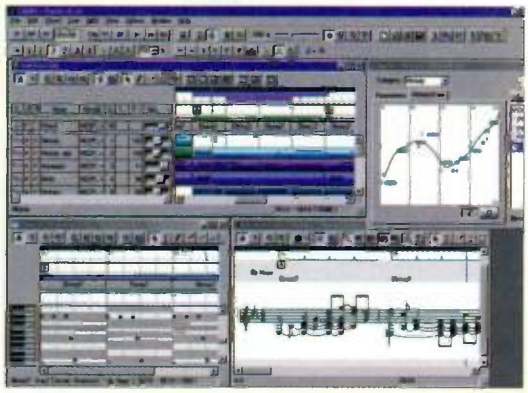
Steinberg North America has added a host of new features and a new look to its *Cubase* series with the release of *Cubase VST* 5.0. *Cubase VST* ($399), *Cubase Score VST* ($549), and *Cubase VST/32* ($799) all offer scalable recording modes, with up to 32-bit floating-point resolution for recording, mixdown, and output in the flagship *VST/32*. *Cubase VST/32* 5.0 also offers a TrueTape mode that uses Steinberg's *Magneto* plug-in technology to simulate the compression caused by tape saturation.

All 5.0 versions of *Cubase* can take advantage of Steinberg's LTB (Linear Time Base) feature, which offers MIDI timing accuracy as precise as 300 microseconds when used in conjunction with the new Steinberg MIDEX 8 USB MIDI interface. The effects section and plug-ins are now easier to use as a result of the redesigned user interface. For example, the new FX Rack offers new controls, and the channel EQs can be edited graphically or with virtual pots.

The InWire Studio feature (with assistance from Rocket Network's Internet technology) allows *Cubase* users to collaborate in real-time in either private or public studios over the Internet. To further aid the collaborative process, *Cubase VST* 5.0 includes the new Universal Sound Module, a GM-compatible virtual sound module that has more than 70 MB of samples.

A new MIDI Track Mixer enables VST-style control over MIDI gear. MIDI editing resolution can be up to 15,380 ppqn; when you use the new Groove Control Browser, the resolution allows for extremely precise positioning of MIDI as well as audio events.

Users can customize and save their window sets, allowing them to call up favorite working setups at any time. A



newly designed Arrange window features Track folders, in which users can store an unlimited number of MIDI or audio tracks. These folders can be shut, which gives you an overview of their contents; stashed out of sight; or nested. With the software's new drag-and-drop features, you can select any musical element—from a single note to a group—and move it from one window to another or to the desktop. Steinberg North America; tel. (818) 678-5100; fax (818) 678-5199; e-mail info@steinberg-na.com; Web www.us.steinberg.net; www.cubase.net.

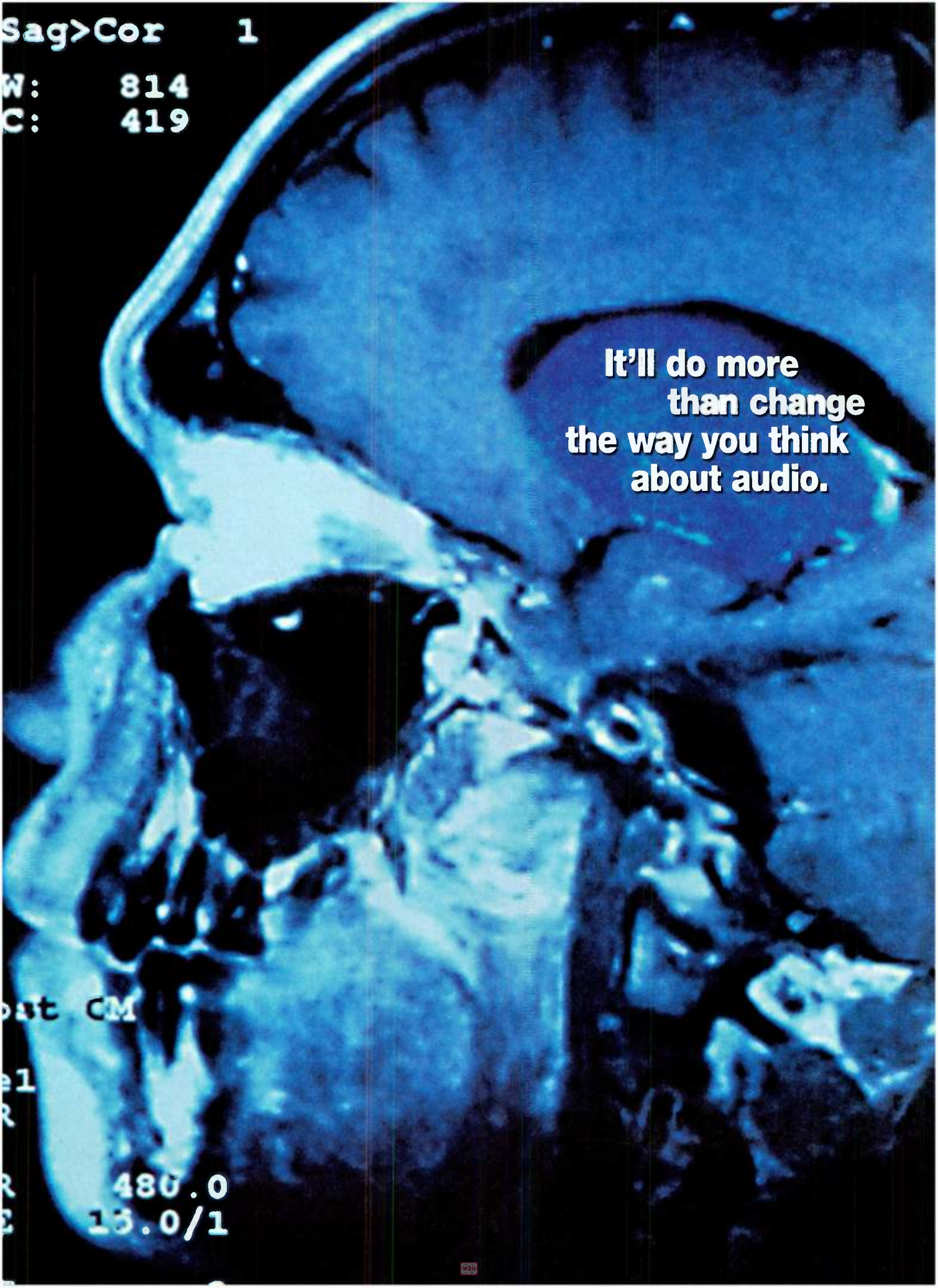
▼ MICROWORKS

Windows users can now take advantage of Microworks *CAMPS* 4.0's sequencing features and compositional assistance. *CAMPS* ($369) combines traditional sequencing elements, such as event lists and graphical MIDI editing, with "intelligent" features that can analyze melodic, harmonic, and rhythmic content and offer musical variations on your theme. *CAMPS* can even generate chord progressions from scratch.

The program can run on any Pentium-equipped PC running Windows 95 or NT 4.0 or later. A minimum of 32 MB of RAM (64 MB recommended) and 80 MB of available hard disk space are required. Microworks Corporation; tel. (973) 492-1691; fax (973) 492-1692; e-mail info@mxw.com; Web www.mxw.com.

▼ BIAS

BIAS's *Peak* 2.5 VST ($499; upgrades from *Peak* 2.x, $99) sports a new user interface and support for VST plug-ins. The new Repair Click function allows you to quickly select and remove a single click. Support has been added for SMIDI transfer to Ultra-Wide SCSI devices, and a more convenient copy-protection scheme eliminates the need for authorizing from a master disk. LE ($99) and TDM ($699) versions are also available. BIAS; tel. (800) 775-BIAS or (707) 782-1866; fax (707) 782-1874; e-mail sales@bias-inc.com; Web www.bias-inc.com.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PROFILE

By Rick Weldon

Love Is a Warm PC

Meg Lee Chin is an American musician currently living in London. For her dance record *Piece and Love* (Invisible Records, 1999), she mixed the beats with streamlined, memorable pop hooks. The music is aimed at the dance floor, but *Piece and Love* originated on Chin's home computer.

Meg Lee Chin gets things done in her London flat.

"Egg Studios is in my living room," she says. "It's basically a PC that I built, a Pentium II/333 MHz running Steinberg's *Cubase VST.* I swipe drum loops from sample CDs, and I flesh out the other parts with [Sonic Foundry's] *Acid* and [D-Lusion's] *Rubber Ducky. Rubber Ducky* is a really simple program that generates bass sounds. While I'm working, I have the program randomize bass lines, and I grab the little bits that sound nice and hooky. I usually import them into [Steinberg's] *Wavelab,* loop them, and put them into a song. On 'Deeper,' for instance, I used a bass line that *Rubber Ducky* created in its Randomize mode." Most of *Piece and Love* is similarly PC based.

Much dance music is the product of the audio tweaker's quest for the perfect original sound, but Chin often relies on immediately usable sounds. She remarks, "Presets are often pretty good. The people making them usually go to the trouble of getting them to sound fairly decent. I know a lot of people reckon that they're better engineers than the people who do presets. But my feeling is that you can use them and get on with making music, or you can spend loads of time messing around, customizing every single thing.

"I like the presets on keyboards, as well," Chin adds. "My friends sometimes say to me, 'Meg, you really ought to learn how to program synths.' When do I have the time for that? I'm writing, I'm playing, and I'm doing video on my computer as well. When do I have time to design my own synth sounds? Why not just quickly get something that sounds good?"

***Piece and Love*/Meg Lee Chin**

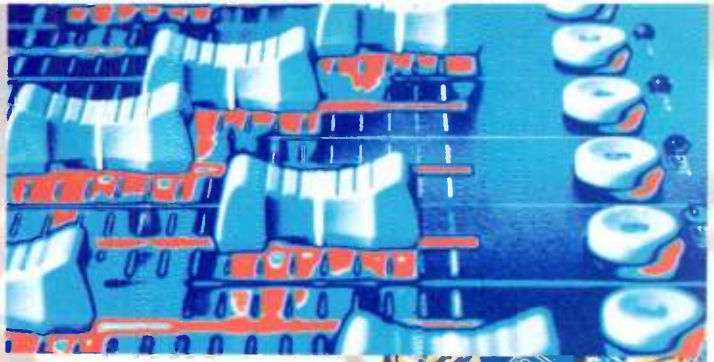
Chin relies on what's at hand to flesh out her songs, but the songs' structures are hardly the product of a formulaic approach. "I'm Miss Versatile—I can do anything," she states emphatically. "A popular Japanese book, *The Book of Rings,* says, 'Never get used to the same weapon,' meaning that you shouldn't always approach things the same way. When I write songs, I can work with a general idea, a melody, a vocal line, someone else's lyrics, someone else's voice, a guitar riff, a breakbeat, a bass line—nearly anything."

Chin feels that she has grown as an artist by kicking the habit of using gadgets and program features to fix problem takes. "When I first got *Cubase,*" she says, "I spent about six months kicking the notes around in MIDI by little increments. It was that long before the lightbulb finally went on in my head, and I thought, 'Well, screw this. I'll just do the take over.'

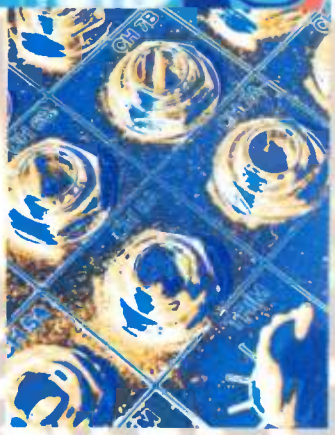
"That's a typical beginner's attitude: trying to fix something with a function key instead of getting a good take. For the same reason, I won't use [Antares's] *Auto-Tune* to fix an out-of-tune vocal. Trying that is usually more trouble than it's worth. It's easier to just redo a track if it's not right."

That approach was helpful in the making of *Piece and Love;* nonetheless, Chin has found that technology can be quite a seductress when it comes to achieving artistic perfection. "Okay, I'll admit it," she adds. "I nudge vocals to make them fit with the beats. I don't *Auto-Tune* them, but I nudge them."

For more information, contact Invisible Records; e-mail egg@yolk.demon.co.uk; Web www.invisiblerecords.com.



When a Tune Hits...



...you need to respond *fast*—before the magic moment is lost forever. So when inspiration strikes, which would you rather do: bust open your computer, wrestle with IRQ conflicts, and troubleshoot a couple of failed driver installations, or hot-plug a full-blown 24/96 audio recorder/processor/controller/mixer into your computer's (or laptop's) external port and get busy making music?

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By Gino Robair

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WEB SITE OF THE MONTH

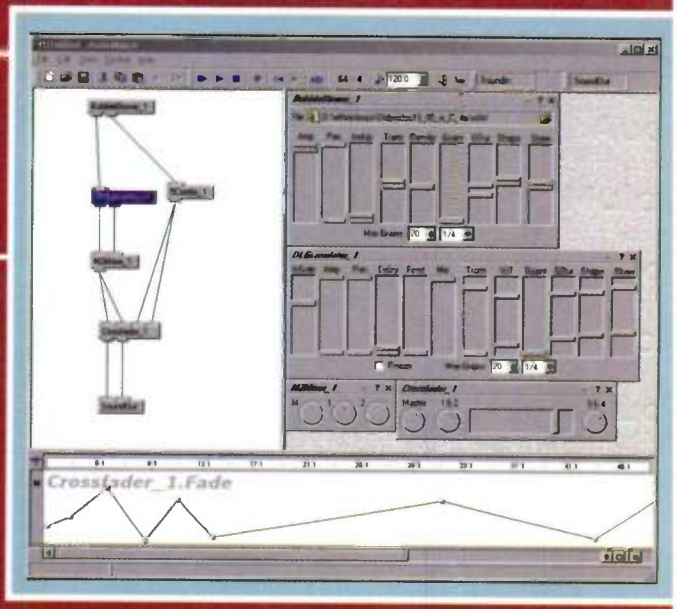
Nowhere is the meeting of art and science so well represented as in the prestigious International Musical Software Competition held in Bourges, France. On its Web site (www.gmeb.fr/SoftwareCompetition/softftpsite.html) you get a g impse into the world of creative freeware and shareware programming. The site lists the prizes awarded in categories such as Multimedia Software, Analysis/Synthesis Out of Real-Time, and Computer-Assisted Composition.

Each of the prizewinning applications has a hyperlink to an explanation and, in some cases, an opportunity to try the software out. I was particularly impressed with the 1999 prizewinner in the Synthesis Software/Real-Time Processing with Gestural Control and Interactivity category, Ryan Francesconi's *Spongefork* 1.04 for Macintosh (http://shoko.calarts.edu/~rf/spongefork). Francesconi describes it as a "self-contained software instrument capable of responding to gestural control while requiring a minimum of hardware components," meaning you get a unique synth and sampler with dual wavetable oscillators that can be controlled by an ASCII keyboard or a mouse. His cleverly designed GUI is a bonus (see **Fig. 1**). The 1996 prizewinner in the category of Analysis/Synthesis Out of Real-Time is well known to readers of **EM**: Tom Erbe's excellent Mac application *SoundHack* (currently in version 0.880).

FIG. 1: Ryan Francesconi's *Spongefork* requires only a keyboard or mouse as a controller.

A.R.T. (www.artroch.com) is accepting submissions for a new Internet Showcase that will be posted on its Web site. The Showcase will include profiles of professionals using A.R.T. products, as well as user tips and advice. The company is requesting press kits and photos to get the ball rolling. For more information, check out its site or send a message to art@artroch.com . . . There are an increasing number of important resources on the Web for songwriters, composers, and other creators and users of musical content. The home page of the **United States Copyright Office** of the **Library of Congress** (www.loc.gov/copyright) includes general information relating to intellectual property and copyright legislation, a tremendously useful FAQ page, and copyright application forms (such as Form PA for registering performing-arts works) in both Adobe *Acrobat* PDF format and as HTML text files. Performing rights organizations **BMI** (www.bmi.com) and **ASCAP** (www.ascap.com) offer useful online forms, info on Internet licensing, music-business features, and a database searchable by song and artist, as well as a variety of information for writers, publishers, broadcasters, and other music professionals. If you need information about licensing or using someone else's compositions, visit the **Harry Fox Agency** (www.nmpa.org/hfa.html), the no. 1 performing-rights clearinghouse. The site has a database searchable by song title, album title, recording artist, or songwriter.

DOWNLOAD OF THE MONTH



Ross Bencina's *AudioMulch* 0.9b2 (www.audiomulch.com) is a shareware application that combines looping and other dance-music features with highly sophisticated real-time processing capabilities. The program is designed to be intuitive to use while giving the user a rich compositional palette to work from.

The various modules in *AudioMulch*—signal processors and sound generators—are referred to as *contraptions*. Sound generators include a drum machine–style sample player, a bass synth, an additive tone generator called 10Harmonics, and a sample granulator called BubbleBlower. The processors include effects (flanger, delay, ring modulation, and the like) and filters (comb filter, parametric EQ, Shepard/Risset filter bank, and so on).

Parameters are MIDI controllable, and *AudioMulch* can sync to an external MIDI clock source. It can process up to 16 channels of audio in real time, depending on the sound card used. Processing can be file based or done live.

You can also use VST plug-ins in *AudioMulch* for even greater processing power. Minimum requirements are a Pentium/100 MHz running Windows 95, 98, or NT (3.51 or later); 4 MB of free disk space; 16 MB of RAM; and a sound card that can handle 16-bit, 44.1 kHz audio.

WEBCAST

If you're tired of listening to MP3s and are looking for a more unique entertainment experience over the Web, live streaming may be the thing for you. An interesting source for Webcasts and chat rooms is Yack.com (http://est.yack.com). Whether you want to hear a live concert, take part in an interview with a recording artist, or explore a non-music-related subject (such as Technology or Health and Family), Yack.com provides a well-stocked repository of streaming content. At the time of this writing, the list of broadcasts included concerts by jazz vibraphonist Gary Burton and rapper Redman and a chat with Grateful Dead drummer Mickey Hart.

Besides broadcasting one-time events, Yack.com includes weekly events such as a forum for drummers, called Talking Drums, and even a show hosted by John Lydon (aka the Sex Pistol's Johnny Rotten). All you need is RealNetworks' *RealPlayer*, Microsoft's *Windows Media Player*, or Apple's QuickTime 4—each is available free.

According to Yack.com, the minimum system requirements for streaming media on a PC are a 80486DX/66 MHz machine running Windows 3.1 (or later) or NT. Mac users will need a PowerPC with Mac OS 7.6.1 or later. On both platforms, you'll need 8 MB of RAM and a connection speed of 14.4 Kbps to use RealAudio, and 16 MB of RAM and a speed of 28.8 Kbps for RealVideo.

WEB APP

Flash Audio

Macromedia's *Flash* is a tool for creating Web-based animation. It uses "vector graphics" to create high-quality, low-bandwidth cartoons that can be posted on the Web and played by any browser that has the *Flash* plug-in installed. The theory is that it's more efficient to send information about how to draw the artwork than to send the artwork itself—from that viewpoint, vector graphics are a lot like MIDI files. Ironically, *Flash* adds movie-style sound to these animations using AIFF and WAV files that have been massively compressed with the MP3 codec. That way, the system can produce interactive audio and synchronized soundtracks in the inherently nonsynchronous environment of the Internet.

Because the cartoons run at different speeds on different machines, platforms, and browsers, *Flash* uses a "keyframe" approach for adding sound. Imported 16-bit, 22 kHz audio files are triggered when specific animation frames are displayed, allowing for both short synchronized sound effects and longer, looping background music. You can also attach sounds to graphic objects and set the trigger to respond to a mouse-click or rollover, providing simple but effective interactive audio capabilities.

Flash also supports an architecture that allows it to communicate with and control other Internet file types. For instance, you can play a QuickTime movie within a *Flash* animation for more traditional synchronized soundtracks or use Lingo-like "FSCommand" parameters to address the powerful sonification tools of the Beatnik technology. (This cross-plug-in connection is called *Flashnik*.)

Even though *Flash*'s internal audio processing is somewhat rudimentary and unwieldy, it does have two distinct advantages over other sound-producing Web applications: first, it contains a very low-bit-rate but high-quality MPEG compression algorithm; and second, it is ubiquitous, making it the de facto standard for audio on the Web.

—Peter Drescher

BAND ON THE WEB

Phish and the Net

The proliferation of online music-delivery sites is making it increasingly easier for consumers to find new and interesting music over the Internet. Major-label recording artists are taking advantage of this by marketing specialty items to enhance their conventional catalog of releases, while fans are fulfilling the promise of the Web by using it to create and foster communities of like-minded listeners. No other group personifies these trends more than Phish.

Originally based in Burlington, Vermont, guitarist Trey Anastasio, bassist Mike Gordon, keyboardist Page McConnell, and drummer Jon Fishman tour continually and have built their following on the strength of their performances rather than through mass-media exposure. Now, after 15 years, these Elektra recording artists have sold an estimated 4 million records worldwide and have legions of Web-savvy fans.

Phish's latest record, *Farmhouse*, is its 11th release on Elektra Records. But the band doesn't let its discography stop there. Phish continually charts high in Emusic.com's weekly Top 10 of the most downloaded music. The online offerings include two full-size releases: *Halloween 1990*, available on a song-by-song basis or in two big chunks (as Set 1 and Set 2); and *Phish* (aka *The White Tape*), a compilation of rare early recordings.

Because Phish allows fans to tape its concerts, there is a rich community of tape traders. Augmenting this are full-time Internet broadcasters of live material, including Phishcast (http://xpager.ndis.umn.edu/phish.html) and Phish Philes (http://phishphiles.livephish.org).

As one might expect, the band's "official" site (www.phish.com) is useful but somewhat commercial in nature—here you can link to a merchandising site known as Phish Dry Goods. However, sites built by "Phish heads" provide the vital grassroots information that fans need, such as weather conditions, housing ideas, and seating diagrams, as well as tape-trading trees, concert set lists, and information on upcoming releases.

One of the more well known fan-made sites is the page offered by Andy Gadiel (www.gadiel.com/phish), which provides tour and ticket info (with links to the venues themselves), as well as Phish-related news items from a variety of media sources, band interviews, and feature stories. Gadiel's excellent Link O' Day page offers additional hours of amusement.

CONSPIRACY BY CONSPIRACY

Live365.com

Internet Radio's "Dirty" Little Secret

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Unformatted and
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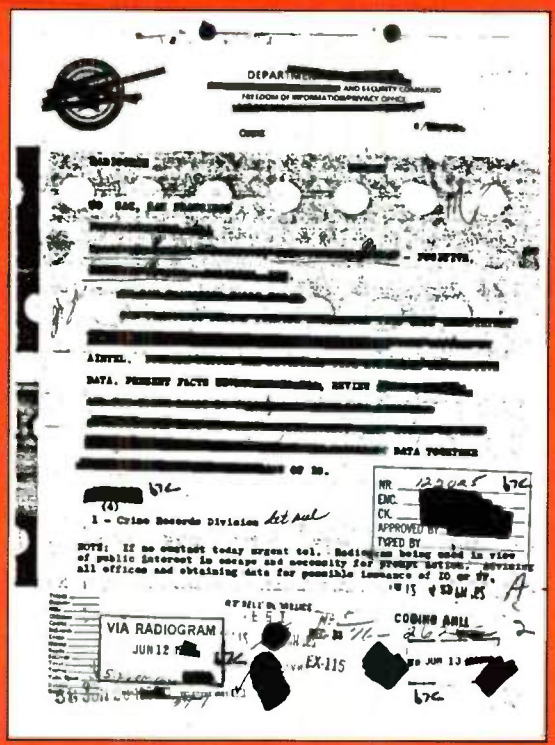
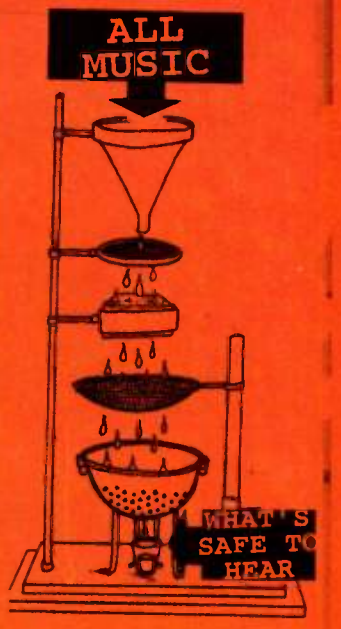
Attention upstanding citizens!!!

Be warned that a new form of radio -- ***digital crack is more like it*** -- is about to be unleashed upon an unsuspecting nation. ***Called Live365.com***, it is nothing less than a plot to lure an entire generation of young hearts and minds into an incredibly hypnotic world of sonic addiction.
Once they've got us hooked, the powers-that-be will start pumping the happy juice into the water supply and using our fillings to manipulate our brain waves.



As this confidential document clearly shows, Live365.com's Internet radio scheme just may be the brainchild of the Freemasons (the government behind the government).



FACT 2

Live365.com is [redacted] to broadcasters and listeners alike. [redacted], can you think of any other place where everything is free? Say, Russia for example!!!!

LIVE365.COM

??????? coincidence??????? Think again, comrades.

FACT 1

Live365.com is many times more powerful than regular radio!!!!!!!!
Regular radio plays the same music over and over and has big blocks of commercials to break everything up. There's no way you can listen to it for any extended period.
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Probability of moral decay??? 100%!!!

FACT 3

***Live365.com begins with the word live and live spelled backwards is **evil**!!!

LIVE EVIL

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Be sure to stuff cotton in your ears (And DON'T look directly at the monitor).

LIVE365.COM™

LIVE365.COM =

Wrong for America

Paid for by the citizens against live365.com

By Scott Wilkinson

Digital Harmony

One of the most vexing problems plaguing studios everywhere is the mass of techno spaghetti needed to get various signals from point A to point B. Each type of signal (analog and digital audio and video, MIDI, sync, and so forth) requires a different type of cable and/or interface, and all signal types must be made to work together as a unified whole. Wouldn't it be great if these signals could be fully integrated and sent over a single cable?

IEEE 1394 is coming to a studio near you.

One of the most promising developments in this regard is a digital interface called IEEE 1394. This high-speed serial interface was first developed by Apple in the 1980s. Apple named it FireWire and submitted the specification to the Institute of Electrical and Electronics Engineers (IEEE), which established it as international standard 1394. Power Mac G4, PowerBook 2000, and blue-and-white Power Mac G3 computers have FireWire ports, and other 1394-equipped products such as hard drives and digital cameras are now widely available.

Devices with 1394 ports can form a network using a variety of topologies simultaneously—daisy chain, star, tree, and so on—as long as there are no closed loops. No more than 17 devices (called *nodes* in network parlance) can occupy a single daisy chain, and the cable length between nodes is limited to 4.5 meters for cheap twisted-pair cable or 100 m for fiber-optic or CAT5 cable. A network can contain up to 63 nodes, and as many as 1,023 networks can be bridged. Each node can include up to 256 terabytes of addressable space.

IEEE 1394 is ideal for media applications in many respects. For one thing, its current bandwidth ranges from 100 to 400 Mbps. A speed of 800 Mbps is expected later this year, and 1.6 Gbps won't be far behind. In addition, 1394 provides *isochronous data transport* for streaming media data. Isochronous data transport guarantees that all nodes have periodic access to the bus, allowing them to send data packets every 125 microseconds. It doesn't provide handshaking or retransmission if an error occurs, but errors are very rare. Even if a packet is lost, 125 µs corresponds to only a few samples of audio data, which is an inaudible gap.

The protocols for sending video data via 1394 are well defined in IEC 61883, a suite of standards codified by the International Electrotechnical Commission. This suite was recently expanded to include an extensible set of audio and MIDI protocols that began life as Yamaha's mLAN (see "Tech Page: Fire in the Wire" in the July 1996 issue of **EM**) and was later adopted by the IEC.

Among the companies working hardest on 1394 media applications is Digital Harmony (www.digitalharmony.com). In partnership with manufacturers of professional and consumer audio/video products, Digital Harmony has developed a licensing program to ensure that products adhere to the IEC 61883 protocols and are therefore completely compatible. The company also wants to be certain that all 1394 media products operate smoothly as a system, shielding the end user from the technicalities of network administration.

Because audio protocols were added to the 61883 suite only recently, no 1394 chips currently include them, so audio manufacturers must build expensive prototyping boards to develop products. Digital Harmony is working with several chip companies on an inexpensive 1394 chip that includes all media protocols. Scheduled for completion early next year, this chip will make it possible to produce reasonably priced 1394 products.

In collaboration with Digital Harmony, Peavey Electronics (www.peavey.com) is developing several 1394-based products for the personal studio (see **Fig. 1**). Such products will provide artists with unprecedented media integration, enabling them to easily complete multimedia projects and publish them on the Internet while banishing techno spaghetti from their studios forever.

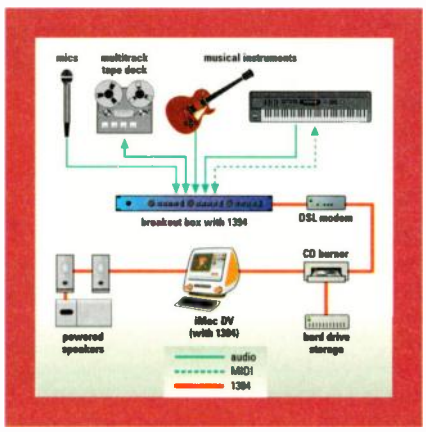

FIG. 1: In a 1394-based personal studio, all data flows from one device to another over a single 1394 cable. Older "legacy" products are connected to the network through a breakout box, which converts various types of signals to and from 1394 format.

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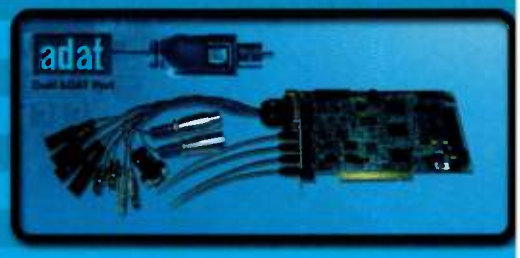

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1.25"(45 mm) silk dome tweeter
Crossover: 2.7 kHz
Frequency Range: 53Hz ~ 22kHz
Normal Impedance: 8 ohms
Sensitivity: 89 dB SPL @1watt, 1meter
Power Capacity: 70W RMS; 120W Program
Dimensions: (H) 380mm x (W) 260mm x (D) 300mm
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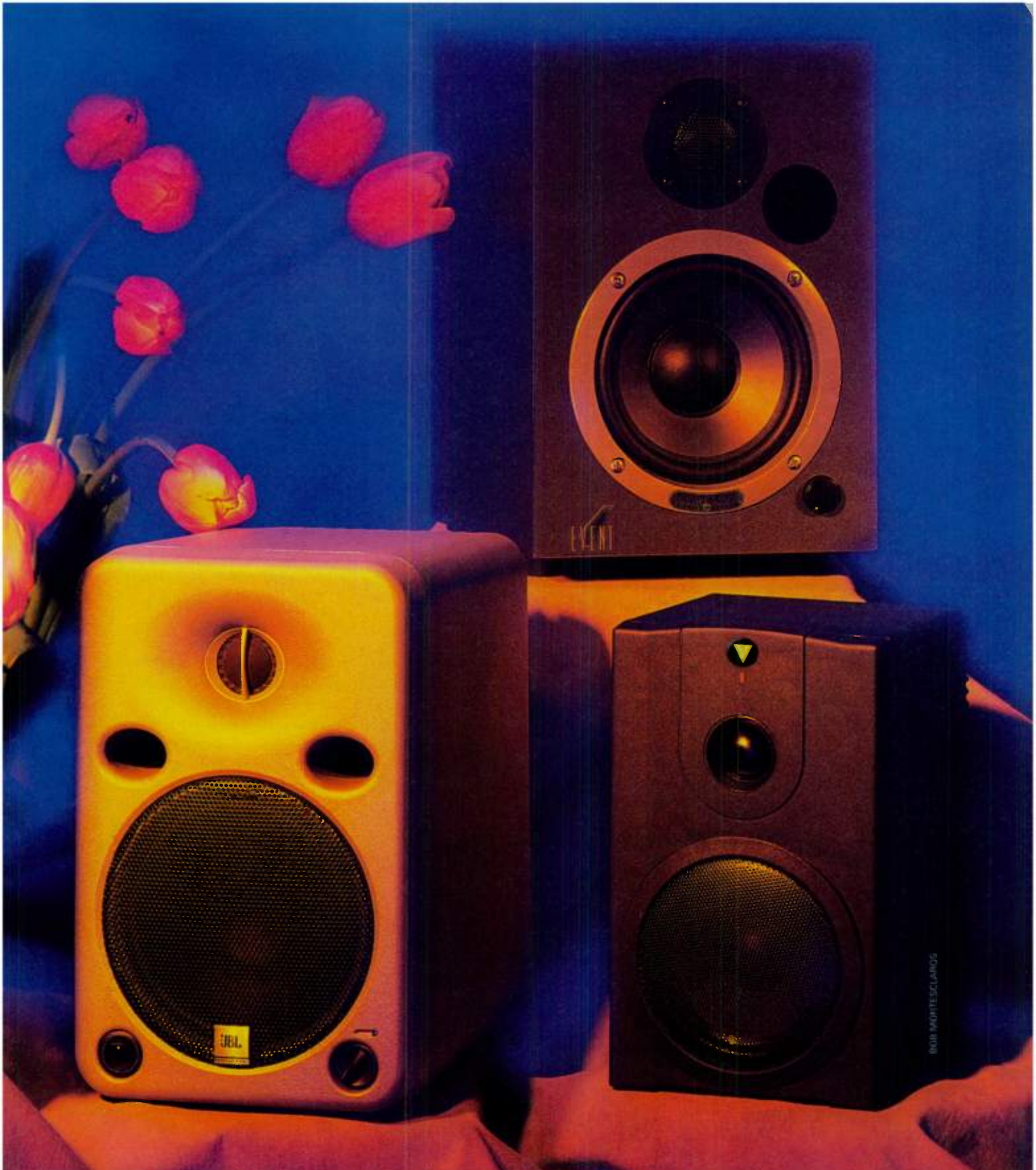
Little Wonders

By Gino Robair

Can you trust what you hear on small active monitors?

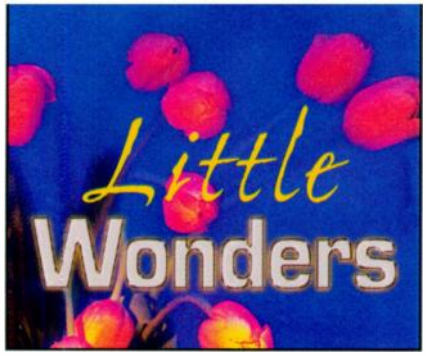
It has been only a few years since affordable powered monitors hit the market in force. Now we're witnessing a similar market surge, but this time the focus is on *compact* powered monitors (defined loosely as models with woofers less than 6 inches in diameter). But wait—we're not talking cheapie multimedia speakers for gamers seeking maximum *boom* while playing *Doom:* Those diminutive drivers have been around pretty much since the multimedia get-go, and have yet to garner heavy raves from any pro-audio folks (for a story on multimedia speakers, see "Little Monsters" in the November 1995 issue of **EM**). The compact monitors under scrutiny here are, in contrast, well engineered, often expensive units from reputable manufacturers of studio reference monitors. They just happen to have wee woofers.

Why are compact powered monitors suddenly all the rage? Largely, it's due to the ascendant popularity of the desktop studio. After all, those who have grown accustomed to having the power of a full-blown studio inside a little box—the personal computer—can only be expected to embrace the "small is beautiful" ethic that pervades the computer market. Moreover, many desktop-studio engineers work in tight quarters (or corners, as the case may be!), so the lighter weight and reduced footprint of compact monitors is doubly appealing. And then there's the issue of



electromagnetic interference (EMI)—those dreaded emissions from conventional, unshielded monitor speakers that make computer VDT screens shake, wobble, and roll.

Of course, desktop studios aren't the only *raison d'être* for quality compact powered monitors. The portability factor makes them attractive also to location recordists, as well as to studio engineers who, in the interest of sonic familiarity, travel with their own monitors. Other potential uses include post-production, surround sound, multimedia, and any application for which compactness is a plus. But for the most part, it's the desktop-studio user whom manufacturers of compact powered monitors are targeting.



QUESTIONS, ANYONE?

For those who are serious about audio production, the proliferation of these miniature speaker systems may cause a certain wariness. It doesn't take an acoustical engineer, after all, to determine that a reduction in woofer size typically leads to diminished low end. Obviously, the big challenge for manufacturers is to get sufficient bass response from such small boxes. To that end, they typically employ equalization and various design features to compensate for the inherent deficiencies of smaller drivers.

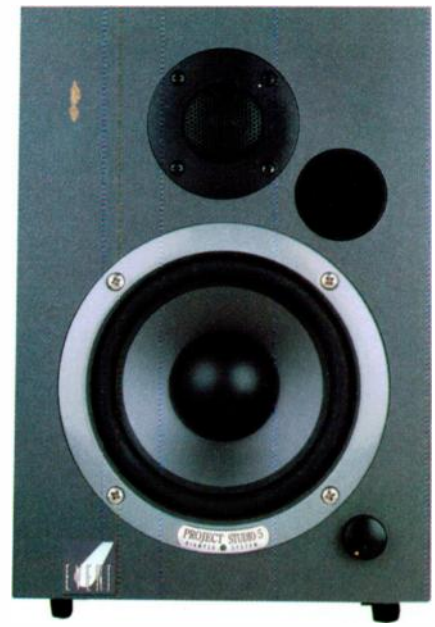
So, have the manufacturers been successful? Is it possible to record and mix with confidence using compact powered monitors? Are some models more capable than others, or do they all fall short of the sonic accuracy required in critical listening applications? In short, what are these speakers really good for?

To help answer these questions, I examined five pairs of speakers with woofers 5.25 inches or less in diameter. I kept within a modest price range: $500 to $1,100 a pair. (Clearly, these monitors are in a class above their consumer-oriented brethren, both in terms of performance and price; you can bet that few gamers are prepared to spend $500 or more for a pair of speakers.) The models I chose were the Event Electronics PS5 ($599), the Genelec 1029A ($1,080), the JBL LSR25P ($958), the Vergence Technology (originally NHTPro) M-00 ($750), and the Yamaha MSP5 ($598).

Interestingly, the number of compact active monitors on the market is quickly increasing. By the time you read this, additional speakers—Hafler's M5 ($498), KRK's V4 ($599) and V4Si ($799), and Tannoy's i5AW ($226)—will have begun shipping. That so many well-respected speaker manufacturers are entering this market is good, if only because it will increase competition and, hopefully, result in better-sounding monitors at lower prices.

COMMON GROUND

To get a feel for the territory, let's first look at features that most of the monitors have in common. Of the five tested, all but one—the Vergence M-00—have woofers measuring 5 inches or more in diameter. (The woofer in the Vergence M-00 measures 4.5 inches.) The M-00 is also the only monitor here that employs a single amplifier—the other four are biamped and employ active crossovers.



Event PS5

Furthermore, the M-00 is the only monitor of the bunch that has neither a volume control nor a bass port. (The purpose of a bass port is to enhance the speaker's low-end reproduction by allowing sound from inside the cabinet to escape.) On each of the other four monitors, both the volume control and bass port—or ports—are located on the front panel.

The volume controls on the Genelec 1029A, JBL LSR25P, and Yamaha MSP5 are stepped, making it easier to match volume levels between the speaker pairs. In addition, the power switches on the Genelec 1029As and the JBL LSR25Ps are located on their front panels, which is a nice convenience.

Note, too, that a couple of the models tested here—the Genelec 1029A and the Vergence M-00—have been on the market for some time. (For a separate review of the Genelec 1029A, see the July 1997 issue of **EM**.)

PROTECTION PLAN

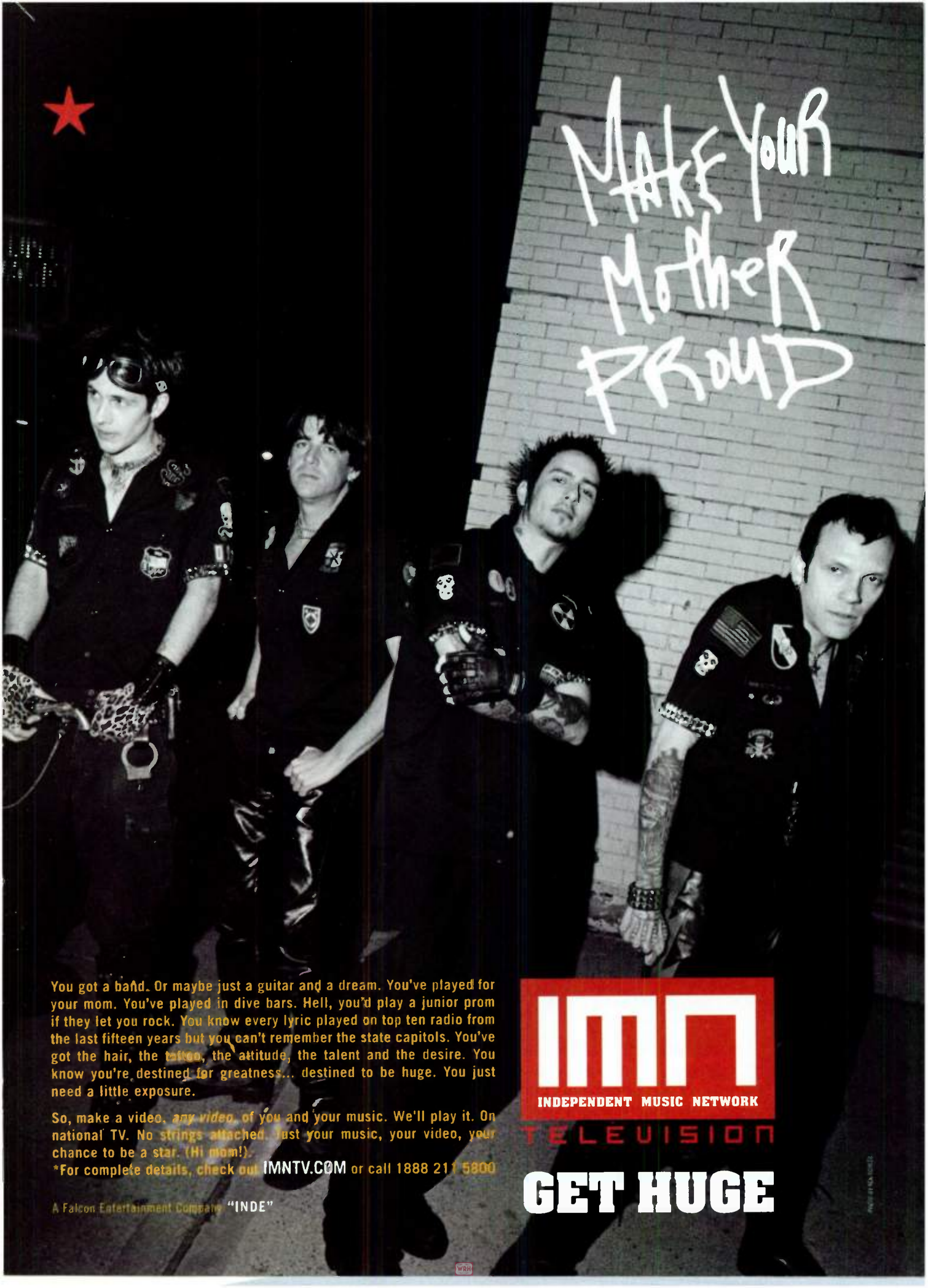
A notable structural difference between the monitors is the implementation of speaker protection—or, in some cases, the lack of it. The Yamaha MSP5s and Genelec 1029As, for example, offer the best protection of both speakers—a metal grille over the woofer and a foolproof "cage" over the tweeter. This is important if you intend to transport

MONITOR PLACEMENT

The placement of monitor speakers in the room has a big impact on the way a system behaves. When your speakers are too close together—such as when they are positioned flush with either side of your VDT monitor—the stereo field is reduced, and reflections from nearby surfaces enhance and cancel portions of the frequency spectrum in complex ways.

Some active monitoring systems include tone controls that allow you to tune the frequency response based on the room and monitor positioning. Moreover, if you've added a subwoofer to the system (often recommended when using compact monitors), you may have to compensate for its subsonic enhancement by attenuating the low end of the satellite monitors. On the other hand, some systems (such as those by Event) allow you to adjust the crossover of the subwoofer to match the natural rolloff of the satellite speakers.

Regardless of where the monitors are in the room, for optimum monitoring it is best to position them so they form an equilateral triangle with the listener. For close-field monitoring—the usual method in personal and desktop studios—this means allowing a distance of roughly 3 feet between the monitors and an equal distance from each of the monitors to your ears. Make sure, too, that there's nothing between you and the speakers. Reflective surfaces in the direct path of the speaker—whether a mixing console, desktop, or the side of a rack case—will degrade the sound.



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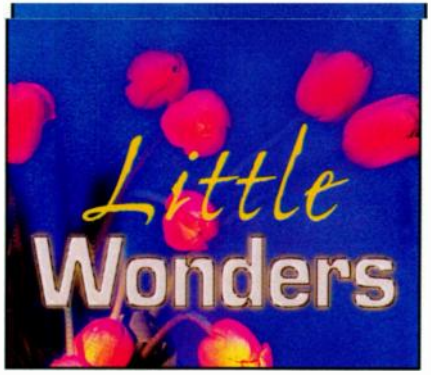

MSP5s are just as tall, but not as wide.) Made of medium-density fiberboard (MDF) and finished in a flat, gun-metal gray, the monitor enclosures, though not unattractive, have a rather plain, no-nonsense look. Unlike the other ported models, each of which has dual, symmetrical bass ports on either side of the speakers, the PS5 employs a single, larger port positioned to one side of the tweeter and woofer. The unit's only front-panel control is a volume knob, which has an orange dot to show its relative position. A green LED beneath the woofer indicates power-on status and flashes at the onset of clipping.

The rear panel of the PS5 acts as a heat sink and provides a power on/off switch, XLR and balanced ¼-inch TRS input (which also accepts an unbalanced ¼-inch TS input), a variable input-sensitivity control, and a connector for the standard IEC power cable (included). The PS5 is the only monitor of the test group that provides no onboard tonal adjustments.

One handy feature, located on the bottom of each monitor, is a tilt bar. This allows desktop users to tilt the front panel 30 degrees upward, thus reducing problems caused by desktop reflectivity while aiming the drivers toward your ears. But even with the tilt bar folded flat (that is, not in use), the feet on the bottom are designed to tilt the PS5 slightly upward.

The PS5s were designed to be suitable for computer-based studio use and, therefore, are magnetically shielded. Unfortunately, the first pair of monitors I received from Event were defective in this regard and caused the image on my computer monitor screen to wobble slightly. However, a subsequent pair provided by Event were adequately shielded.

Genelec 1029A

Sonically, the PS5s are quite smooth from bottom to top, although I did notice a bit of tubbiness in the low end on some instruments. The transient response is above average. Percussion sounds great through these monitors, as do vocals and electric guitars. Overall, the PS5s sounded warm and well balanced, and were pleasant to listen to.

Genelec 1029A. The most expensive monitors of the group, the Genelec 1029As are in many ways also the most versatile. They provide four rear-panel tone-control DIP switches, allowing you to customize the response to the acoustic environment—an especially useful feature for surround-sound applications. The DIP switches, though a bit of a hassle to deal with, provide a fair amount of tailoring. Switch 1 is the "treble tilt," which attenuates the high end by roughly 3 dB, beginning at 4 kHz. Switch 2 is a bass-rolloff shelving filter positioned at around 90 Hz. (Theoretically, this switch would be engaged only if the Genelec 1091A subwoofer were part of the setup.) Switches 3 and 4 cover "bass tilt" duties, allowing low-end cuts beginning at approximately 1 kHz. (The bass-tilt controls also cut a dB or two from the spectrum between 1 and 3 kHz.)

Except for the Vergence M-00, the 1029A is the smallest of the five monitors. The enclosure is cast aluminum

WHY POWERED MONITORS?

Powered monitors are essentially plug-and-play devices: just insert the power cable into an AC outlet, feed the monitor a line-level signal, and you're ready to roll. Some powered monitors have a single amplifier that powers both woofer and tweeter. Others, called *biamped* monitors, employ two separate amps, one each for the woofer and tweeter. A three-speaker system with an amplifier for each speaker is referred to as a *triamped* system.

However, active monitors are more than just speakers with dedicated amplifiers. Besides the drivers, power supply, and amplifiers, powered monitors also include, among other things, crossovers that determine where the high and low frequency range is divided between the tweeters and woofers. The benefits of having all of this equipment on board include better transient response, larger dynamic range, and improved phase coherence.

Powered monitors solve a number of other problems for the engineer. To begin with, they offer a complete and optimized listening system. There's no need to worry about compatibility between speakers and power amps, because each component in the system is carefully matched for optimum performance. And, because active monitors are self-contained, they're easy to travel with. It's not uncommon for an engineer to show up for a session at an unfamiliar studio with his or her favorite powered monitors in tow.

My first experience with powered monitors was not in the confines of a studio control room, but in a cathedral where the engineer was recording a pipe organ. Rather than rely on headphones alone for monitoring, the engineer brought a pair of powered monitors and set them up in a back room that was sonically isolated from the sanctuary. That way, he was able to get a much truer sense of how the recording would translate to playback. And, by bringing active rather than passive monitors, he was able to leave his power amp and extra cables behind.

For manufacturer information, please see Contact Sheet, p. 205.

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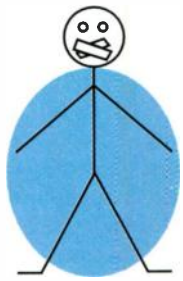
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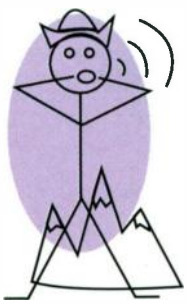
Hiss Reduction


Distortion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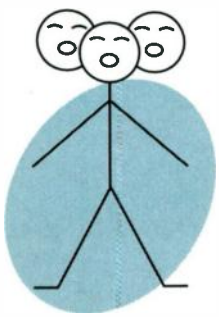
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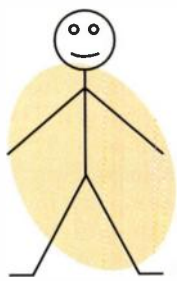
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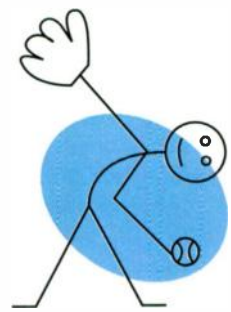
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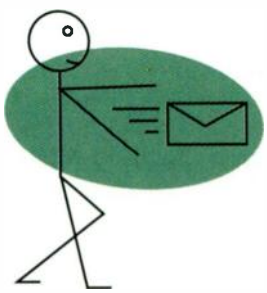
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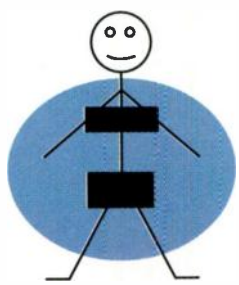
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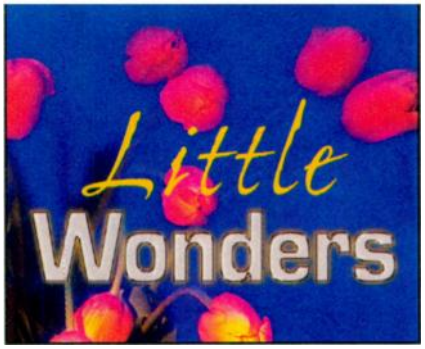

and provides good electromagnetic shielding. Finished in black semigloss (and available in white or gray, too), the units have a simple yet elegant look. The slitlike ports, located on either side of the tweeter, are flush with the sides of the cabinet, and the power switch and volume control reside beneath the woofer on the left and right sides, respectively. A green LED located above the power switch indicates power-on status.

The 1029A's rear panel, which doubles as a heat sink, provides simultaneously available, balanced XLR and ¼-inch connectors; recessed DIP switches; a recessed voltage selector; a connector for an IEC power cable (included); and predrilled holes for mounting the unit on an Omnimount Series 50 bracket. There is also a ⅜-inch threaded hole on the bottom of each monitor that can accommodate a standard microphone stand (which is handy if you follow Genelec's suggestion that the monitors *not* be positioned on a console or desktop).

The frequency response of the 1029As is fairly even between 100 Hz and 1 kHz. There are, however, noticeable boosts between 70 and 90 Hz and between 1 and 4 kHz. The low-end boost was the more problematic of the two—with the speakers set "flat" (all switches in the off position), drums and bass tended to sound muddy, especially when playing together in a mix. After a bit of experimentation, however, I found the best DIP-switch settings for my room, which included using switch 2 (the 90 Hz bass rolloff) to reduce the annoying low-end bump without sucking out all of the bass frequencies.

The 1029As have a perky high end, but I didn't find it bothersome or fatiguing. Rather, it provided natural-sounding detail and air that were missing from most of the other monitors. The stereo imaging was good, and on acoustic instruments, including a full-blown jazz-horn section, the tonal balance was remarkably smooth. Moreover, vocals sounded so good through the 1029As that I would guess these monitors were created with the human voice in mind.

JBL LSR25P

JBL LSR25P. Like the 1029A, the JBL LSR25P Linear Spatial Reference biamplified monitor provides both a front-panel power switch and volume control (also positioned beneath the woofer on either side), indented wave guides around the tweeter, and rear-panel DIP switches for tuning the monitor. However, at 17 pounds each, the LSR25Ps are much heavier than the Genelecs, and they also sound remarkably different.

Made of cast-aluminum with a silver-gray, semigloss finish, and sporting oval bass ports and the aforementioned vertical tweeter-protection bar, the LSR25P is decidedly more space-age-looking than the other monitors. The snazzy vibe is further enhanced by the shape of the cabinet, which flares out from back to front, and is quite

A Look at the Manufacturer Specifications

	Event PS5	Genelec 1029A	JBL LSR25P	Vergence M-00	Yamaha MSP5
Woofer Size	5.25"	5"	5.25"	4.5"	5"
Tweeter Size	1"	0.75"	1"	1"	1"
Power Amp Output, Woofer (RMS)	70W	40W	100W	75W	40W
Power Amp Output, Tweeter (RMS)	30W	40W	50W	not biamped	27W
Peak Output (SPL)	104.5 dB	110 dB	108 dB	111 dB	101 dB
Inputs	(1) XLR; (1) ¼" TRS	(1) XLR; (1) ¼" TS	(1) XLR; (1) RCA	(1) XLR; (1) ¼" TRS	(1) XLR; (1) ¼" TS
Input Sensitivity	+4 dBV/–10 dBu	+4 dBV/–10 dBu	+4 dBV/–10 dBu	+4 dBV/–10 dBu	+4 dBV/–10 dBu
Frequency Response	53 Hz–19 kHz (±3 dB)	70 Hz–18 kHz (±2.5 dB)	70 Hz–20 kHz (+1/–2 dB)	98 Hz–20 kHz (±2 dB)	50 Hz–40 kHz (–10 dB)
Crossover Frequency	2.6 kHz	3.3 kHz	2.3 kHz	2.2 kHz	2.5 kHz
Signal-to-Noise Ratio	>100 dB (unweighted)	>90 dBA	>98 dBA	95 dBA	100 dBA
THD+N	<0.01%	<0.08%	<0.05%	<1%	0.02%
Price per Pair	$599	$1,080	$958	$750	$598
Dimensions	7.5" (W) × 10.5" (H) × 9" (D)	8" (W) × 9.75" (H) × 5.63" (D)	6.8" (W) × 10.6" (H) × 9.5" (D)	5.7" (W) × 9" (H) × 7.3" (D)	6.5" (W) × 11" (H) × 8.7" (D)
Weight (ea.)	16.5 lbs.	12.5 lbs.	17 lbs.	14 lbs.	16.5 lbs.

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The LSR25P's rear panel, which also doubles as a heat sink, provides an unbalanced, –10 dBu RCA jack; a balanced, +4 dBV XLR jack; four recessed DIP switches; and a connector for the IEC power cable (included). There are also predrilled holes for mounting the unit. Curiously, the XLR jack is nonlocking, which is a bit disconcerting when you consider that it's mounted vertically. DIP switch 1 is an 80 Hz high-pass filter, switch 2 is "boundary compensation" (a low and low-mid cut), and switches 3 and 4 let you boost or cut, respectively, high frequencies by 1.5 dB from 2.2 kHz upward.

Impressively, the LSR25Ps sound great right out of the box. In terms of full-range frequency response, they were the most evenly balanced of the bunch, producing a realistic sound even on demanding instruments such as classical guitar and grand piano. The bass and low-mid frequencies seem slightly enhanced (again, with the tone switches off), which helps the monitors maintain full lows even at lower volumes. However, the low boost isn't excessive (it didn't, for example, register as much rumble from a floor tom or bass guitar as did most of the other monitors), which is good because it keeps the LSR25Ps from sounding artificial.

I also noticed what seemed to be slight dips at around 3.5 and 12 kHz. Overall, though, the upper frequency range is clear, distinct, and pleasing to the ear. Transient response, too, is exceptional. I got used to this pair of monitors very quickly.

Vergence M-00. Though weighing in at only 14 pounds each and employing the most diminutive woofer of the group, the Vergence M-00s pack quite a sonic punch. The cast aluminum/zinc body makes this not only a durable unit, but also provides excellent shielding. The finish is flat black and the look understated.

All M-00 controls, including the main power switch, are located on the unit's back panel. There are three switches. The Position switch lets you choose between NF (near-field, or up to three feet from the speakers) and MF (mid-field) settings. This is the only EQ adjustment available. The NF setting subtly attenuates upper frequencies, beginning at 3 kHz and sloping to a maximum of 3 dB attenuation at 20 kHz.

The second rear-panel switch, which can be set to –10 dBu or +4 dBV, controls input sensitivity. The M-00 is the only speaker in the bunch that offers three different connectors: XLR (balanced), ¼-inch (balanced or unbalanced), and RCA.

The third rear-panel switch controls a power-saver feature: in Auto position, the speaker will go into a low-power, standby mode if no signal is detected for approximately ten minutes. This feature allows you to leave the main power switch on. (You can also leave the switch in the On position and use the main AC switch to power the speakers up or down.)

Compared with the other monitors, the M-00s had a tight, somewhat claustrophobic, and slightly two-dimensional sound quality, with a noticeable muddiness in the low and low-mid frequencies. In addition, the M-00s sounded "throaty" in the midrange, which was especially noticeable on piano tracks. The upper range was better, but sounded slightly diffuse and unfocused.

Vergence M-00

Yamaha MSP5

The M-00s came off sounding more like home-stereo speakers than reference monitors when compared with the other units. Of course, this could be a plus if you need a secondary speaker pair to test mixes on a "real-world" listening system. In addition, the M-00s were the noisiest pair that I tested: they pop when turned on and off, they pop going in and out of standby mode, and they emit more hum than the others. In standby mode, however, the M-00s are practically silent.

Yamaha MSP5. Despite a relatively modest power output maximum of 40 and 27 watts for the woofer and tweeter, respectively, the Yamaha MSP5s proved easily the loudest pair in the group. They are also well shielded and, as mentioned previously, feature full protection for both tweeter and woofer (which, come to think of it, is a big plus if your desktop studio is within reach of children!). The plastic cabinets have a flat-black finish and a slight downward slope—a useful design if the monitors sit above your desktop. In my case, however, with the monitors roughly at ear level, I needed to elevate them slightly in front.

The MSP5s have a straightforward, down-to-business sort of look, with two circular bass ports situated between the

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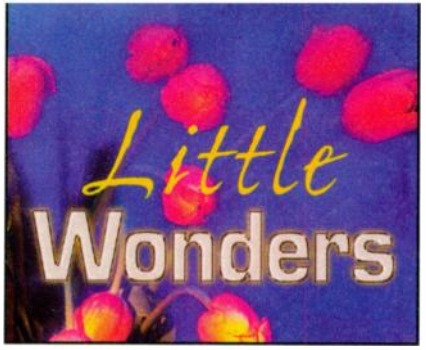

tweeter and woofer, one on either side. The stepped volume control is located in the lower right corner and a green status-on LED in the lower left.

A large, attached heat sink takes up much of the rear panel. The power switch is located below the heat sink, along with unbalanced ¼-inch (–10 dB) and balanced XLR (+4 dB) input jacks. There is also a pair of frequency-trim switches labeled "Low" and "High." The Low switch has four positions: in addition to the 0 setting, you can either add 1.5 dB or subtract 3 dB in two 1.5 dB increments at around 60 Hz. (These changes affect the frequency band from 400 Hz downward.) The High switch has three positions, allowing you to add or subtract 1.5 dB from the 0 setting at around 25 kHz. (This cut reaches down to the 3 kHz range.) Unlike the other monitors in the group, the MSP5 has a permanently attached power cable.

The MSP5s have a robust low end, scooped-sounding lower-mids, and a brittle high end—much brighter than the frequency-response graph would suggest, with noticeable boosts between 3 and 10 kHz. There is also a noticeable bump in the low end, around 80 Hz. In terms of frequency response, I found the MSP5s to be the most "enhanced-sounding" of the units tested, and hence the least reliable as critical reference monitors.

Fortunately, the frequency-trim switches allowed me to attenuate the boosts somewhat. I elected to cut both the highs and lows by 1.5 dB. Actually, I would have cut the highs further had there been the option. These monitors are crispy sounding, and probably the most ear-fatiguing of the bunch.

I must admit, though, that certain things sounded great on them. Heavy rock mixes, for example, seemed to jump out of the speakers. The MSP5s also delivered the best kick drum and floor tom sounds of the test group. Moreover, there was clear delineation between the drums and bass. As fun as they were to listen to, though, the MSP5s didn't instill much confidence in terms of reproducing a balanced frequency range.

A bass port enhances a speaker's low end.

WONDERS NEVER CEASE

Though each of these monitor pairs made for a good listening experience, after working closely with them, I can only conclude that small speakers—those 5.25 or less inches in diameter, at any rate—are inadequate as woofers. They simply cannot compete with larger speakers when it comes to smooth and accurate low-end reproduction. Therefore, I wouldn't choose any of these models as my primary reference monitors, at least not without the addition of a subwoofer. (Note that three of the models tested here are available with system-matched subs: the Event Electronics PS5s, the Vergence M-00s, and the Genelec 1029As.) With time, of course, one can learn to mix well on practically any monitors. But if you are looking for an accurate critical reference monitor for your desktop studio, I would suggest that you stick with one that features a standard-size woofer—something at least six inches in diameter.

Of course, that's not to say that the five monitors tested here are inadequate for other applications. On the contrary, I would welcome any of them as a secondary pair of speakers in a mixing session. Moreover, any of them would make killer multimedia speakers, and most would probably also prove impressive in home-theater surround-sound applications.

In terms of overall tonal balance, I found the Event PS5s, Genelec 1029As, and JBL LSR25Ps most trustworthy. Therefore, of the five, these would be my first picks for location recording and post-production. If size is a primary consideration, the Vergence M-00s and Genelec 1029As deserve a close look. And if what you need is a rocking speaker with lots of volume—for intensive multimedia, say, or installation in the game room—the Yamaha MSP5s will fit the bill and then some.

Gino Robair *is an associate editor for* **EM.**

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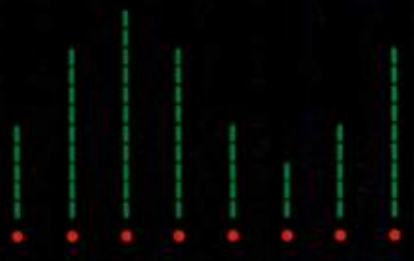

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One of the software categories that has benefited most from recent advances in computing power is software synthesizers. Today, turning your computer into a full-blown synthesis engine is nearly a trivial matter, and even systems that are less than state-of-the-art can usually provide valuable new resources to the desktop composer.

This article will examine a wide range of programs that use your computer's CPU as the sole sound-generating hardware, often with all the adjustable parameters that you'll find in today's outboard synthesis gear. I'll look at the most recent versions of some older programs and discuss several of the hottest new releases on the market.

Because so many programs are available, I'll only cover a few in each of the main categories. No doubt you'll find many other excellent programs, especially shareware, by surfing around the Net. Also, my focus will be on Macintosh and Windows applications, but you can certainly find high-quality options for other platforms such as Linux and BeOS. And I'll discuss only programs that run free of any dedicated hardware, which means that plug-ins written for TDM, for example, won't be included.

SOFT

by Dennis Miller

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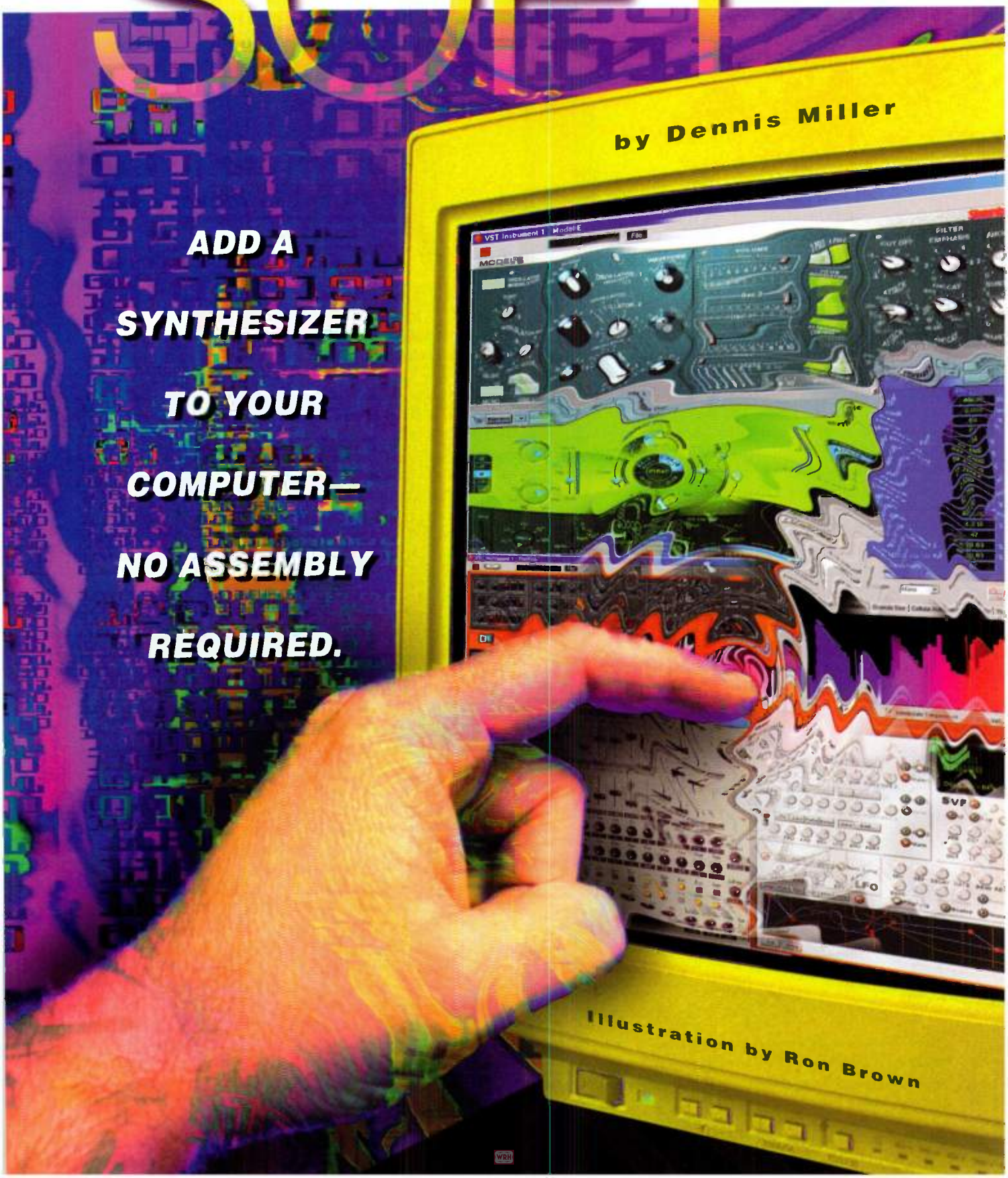

Illustration by Ron Brown

THOSE WERE THE DAYS

The concept of native signal processing, in which the computer's CPU provides all the computing power that a real-time multimedia program needs, is not new. In fact, it was originally brought to market by Intel, Motorola, and others in the early to mid-'90s—alas, somewhat before its time. Only in the past few years, with the advent of vastly more powerful processors, has it been possible to realize the potential of native signal processing, and all of these programs operate under that basic process.

Of course, when Max Mathews at Bell Telephone Labs first developed the means for computers to make sound in the late 1950s, the method of specifying values for sound parameters was much different from what it is today. Mathews and his colleagues used punch cards to enter their scores, as well as the routines that the computer used to generate sound. That method remained current for several decades, until terminals on which data could be typed and sent into the computer became available. (See the sidebar "Other Avenues" for a description of two modern programs that stem from Mathews's work.)

What a long way we've come. The soft synths covered here offer a host of real-time controls, including onscreen sliders, knobs, and faders. They also can be controlled from external MIDI keyboards, joysticks, and the like, allowing you to "play" them live. And in most cases, you can control them from a sequencer running on the same computer.

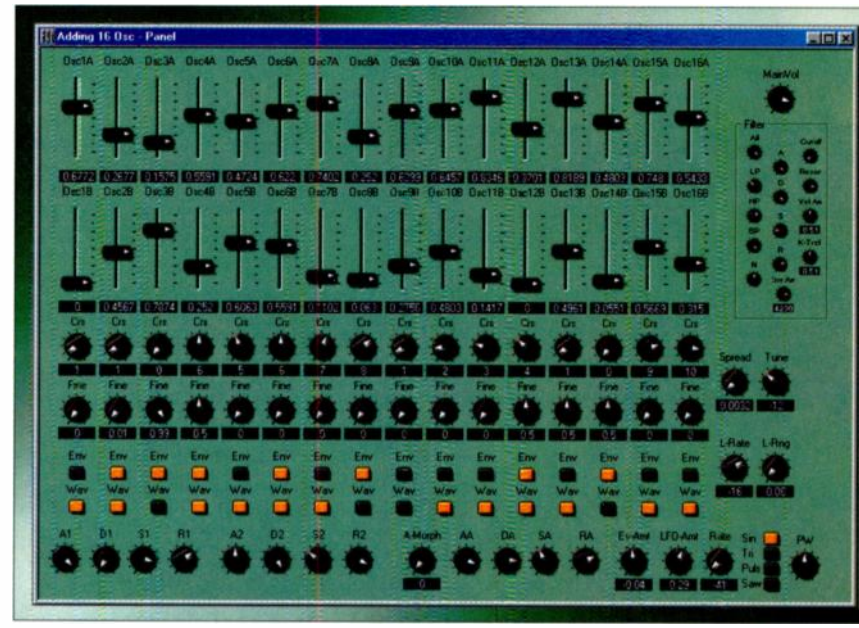

FIG. 2: *Reaktor* provides dozens of modules with which you can build elaborate synth designs. The synth pictured here uses additive synthesis with filters and envelopes to create sound.

YOU PAYS YOUR MONEY

What are the advantages of using a soft synth? Sound quality comes to mind; if your audio hardware supports digital I/O and has multichannel outputs, you can look forward to excellent audio quality with considerable routing flexibility. Another advantage is independent effects processing on each MIDI channel, a capability you'll get from several of the multitimbral synths. With so many basic design components available, you can also expect a high degree of sound-design flexibility; and of course, adding new features by way of software downloads is a cinch.

But there's no question that dedicated synthesis hardware also has its advantages. These include reliable and consistent performance (because it's not dependent upon a host CPU), portability (rack it and take it on the road), low or nonexistent latency, and ease of real-time tweaking (just how many parameters can you change at once with a mouse?). Nevertheless, a synthesizer running completely in software can be the best solution for many musicians, and it will certainly be a tremendous asset even if it's not your only means of creating sound.

By the way, I won't be making any absolute judgments about performance here, because it varies greatly depending on many factors (see the sidebar "Performance Practices"). There are simply too many variables to establish any relevant benchmarks.

However, on the test beds that I used—a Pentium II/400 MHz with 128 MB of RAM and a Mac G3/266 MHz with 256 MB of RAM—I got completely satisfactory performance from all these programs when using them as my only sound source. (Using a soft synth while playing audio tracks in a sequencer or attempting to sync incoming MIDI data with events in a sequencer track being routed to a soft synth can be more problematic.) I'm confident that a soft synth can offer valuable sonic resources on any modern machine; but, of course, the more horsepower you can put at your synth's disposal, the better your results will be.

FIG. 1: *Tassman* offers a number of physical models among its many synth modules. Pictured here is a synth that contains a pick, string, and metal plate, as well as a sequencer and several modulation sources.

SYNTH CONSTRUCTION SETS

Dozens of software synthesizers are available today, and for the most part they fall into several basic categories. The first group that I'll examine consists of

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programs that are, in effect, synth construction kits. They offer building blocks for creating your own sound modules, and one gives you tools not yet available in any of the most modern hardware synthesizers. The synths in this group run as stand-alone applications, but some also work as plug-ins, and most are intended to be operated live using MIDI note and controller data or to be triggered by a sequencer.

These programs share other attributes. For example, most, if not all, can write their output directly to disk, giving you a 16-bit pure (or better) signal path straight to your CD burner. (Many programs in other categories have this same capability.) The majority of the synth construction kits have some type of built-in sequencer for triggering sounds and accepting MIDI input. And though you'll find significant differences among the interfaces that they offer, each has some form of modular, "patchable" design layer in which you connect various components to create your final, unified design.

Here, then, is a look at several programs that provide a modular approach to sound programming with real-time MIDI control.

***Tassman* 1.2 (Win; $395).** One of the most interesting new modular synths is Applied-Acoustics' *Tassman*. Other synths have used physical modeling in the past, most notably Seer Systems' *Reality*, but *Tassman*'s catalog of sound-producing options is unique. Here you'll find bows, bars, beams, and many other objects with which to build virtual instruments (see Fig. 1). These objects open up a vast realm of sonic possibilities that you won't find in any of the other programs reviewed here.

Tassman splits its functions into two independent applications: *Builder*, with which you design and test your synths; and *Player*, which you use to play them. This is a bit unusual and somewhat unwieldy; although you can load *Player* from within *Builder*, it would be easier to have a single screen from which you could load any of the dozens of presets, play them, and modify their designs. Instead, you typically access synths directly from an Explorer window, then choose whether to run them in *Player* or load them into *Builder* for modification.

Tassman's sound-design tools are extensive but not as numerous as those in some of the other programs. On the other hand, *Tassman*'s unique physical-modeling modules really give the software its identity. Interested in the sound of a marimba being bowed or a flute being struck by a mallet? In *Tassman*'s world of virtual instruments, these and many other anomalies are possible. Moreover, the parameters that the modules use, such as pick stiffness and mallet strength (force), are intuitive and easy to control.

Applied-Acoustics gets high marks for its documentation, which includes a well-paced tutorial and a good explanation, along with suggested usage tips, for each of its modules. The coding also seems very efficient; I was able to achieve high levels of polyphony with many of the presets. If you're looking for some new sonic dimensions to add to your work, *Tassman* could be just right. It's one of the most exciting new programs that I've seen in a while.

***Reaktor* 2.3 (Mac/Win; $499).** Native Instruments' software is noted for its power and flexibility. The company's latest offering is *Reaktor* 2.3, which combines the synth application formerly known as *Generator* with the company's sampling workhorse, *Transformator*. (These products are no longer available separately.) This potent combo provides a massive array of sound-generating and sound-processing modules that you can freely connect to build your synth designs. A huge number of presets are also included, created by both the company and *Reaktor* users.

Reaktor's synthesis options run very deep. To access any of the dozens of modules, you open a new Ensemble and insert the components that you need. Modules come in various levels of complexity: some are macros that perform several functions; others are more basic and can handle only a single task. The toolkit includes functions for many different mathematical operations and for more familiar sound-generating processes.

Once you have completed an Ensemble, you switch to the Panel display in order to play it and adjust its parameters (see Fig. 2). In the Panel, you can take snapshots, which capture the status of all the parameter values at a given moment. Snapshots can be accessed from a drop-down menu and via Program Change messages.

Reaktor's sampling features are well integrated with its synthesis functions and are a huge resource in their own right. The combination of a vast array of sound-generating modules and numerous sound-transformation features makes *Reaktor* an excellent choice for a wide range of tasks. Its ability to function as a VST Instrument is simply icing on the cake. And if you get the free Premium Library (an excellent collection of presets), you'll be ready to run dozens of patches right out of the box.

***Buzz* 1.2 (Win; free).** Oskari Tammelin's *Buzz* is a modular synth builder, but it looks and feels quite different from most of the programs in this roundup. It appears to enjoy a wide following; many of the bundled presets were created by users, and the Web site is an active spot for exchanging information about the software and related topics.

There are two main categories of synth-design modules: Generators and Effects. *Buzz* calls these modules *machines*, and you build complex synth networks by wiring the machines together. To control the machines, you create Patterns that remain fixed or vary over time. Many (but not all) machines can also employ MIDI input either alone or as a "live accompaniment" to the

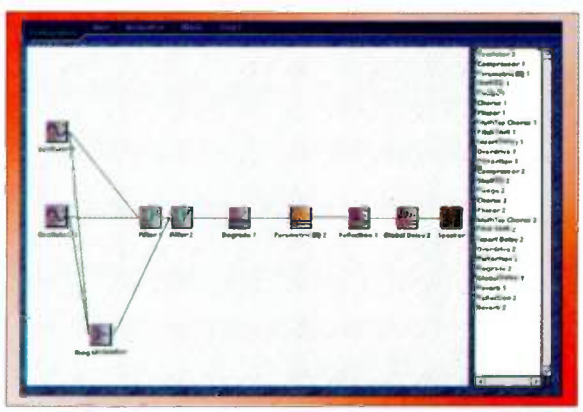

FIG. 3: *Retro AS-1*'s Configuration window presents a graphic overview of your synth design. Synth parameters are edited in several of the program's other windows.

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control data contained in the Patterns.

Patterns are created in the Pattern Editor and then scheduled in the Sequence Editor. The Pattern Editor is, in effect, a sequencer that can be used to control the time-varying evolution of any machine parameter. This means that you can animate the wet/dry mix of a reverb, the depth of a flanger, or simply the frequency of an oscillator. Tools to help you tweak your Patterns are provided; for example, the Interpolate function creates a series of values from a given start point to an end point. The Randomize feature alters the values in a range. Many of the values that you supply for your Patterns must be hexadecimal, so get out your calculator if you're not comfortable with base 16.

You can insert any number of Patterns into the Sequence Editor and then set the Editor to one-shot playback or make it loop over a selected range. You can adjust the tempo as a machine plays and view a display that continuously updates the value of any chosen parameter. A CPU monitor allows you to see how hard your computer is working during playback, and you can also watch a real-time frequency analysis of the audio being generated.

Buzz seems to play well with others; it can load VST plug-ins as effects, and there's even a browser plug-in available that lets visitors to your Web site hear your *Buzz* creations. You can also load *Buzz* files into Nullsoft's *Winamp* using a free plug-in. *Buzz* allows you to build your own basic synth components, but you'll need some programming chops and a Visual C++ 5.0 compiler for the task. However, the overall flexibility of this program could make it a major player in your audio environment.

***Retro AS-1* 2.0 (Mac/Win; $259).** BitHeadz has recently released a major upgrade to its *Retro AS-1* software, and this is one of the most potent tweaks in memory. There are numerous enhancements; for example, version 1.0 offered a maximum of 32 voices, but the new 2.0 bumps that up to 64. The company has also added new filter types, new insert effects, and significant enhancements to the interface. You can even route external audio into the program and use it as a modulator in the new Ring Modulator section. As a bonus, *Retro AS-1* is fully ReWire-compatible. (A separate program version—*Retro AS-1 VST*—can be used as a VST Instrument on the PC.)

In *Retro AS-1*'s Configuration window, you view and add or remove modules in your synth design (see **Fig. 3**). Clicking on any module brings up the Main window, where you can set values for any of the module's parameters. Mod sources and effects (up to two insert and two global per patch) are configured in their own respective windows. In the final screen, Global, you choose from various voice-activation, portamento, and MIDI-related options. (It takes a bit of window jumping, as you might imagine.)

Each of *Retro AS-1*'s three stereo oscillators can use one of ten waveforms or three types of noise. For those who really like to drive their sounds hard,

OTHER AVENUES

Though all of the programs I've covered here display their features on graphical user interfaces, that's not the only approach to generating sound in real time. Two sound-programming languages, Barry Vercoe's Csound (free) and James McCartney's SuperCollider ($250), also have the ability to create and modify sound as it plays back. Both programs let you send MIDI data from a controller to manipulate sound parameters, and SuperCollider can also incorporate graphic control panels containing sliders or buttons into its patches (see **Fig. A**).

With programming languages such as these, you build "virtual instruments" by accessing the huge libraries of functions that each language provides. In most cases, this involves typing text into a word processor or text editor (one is included with SuperCollider) and then telling the programs to compile or "render" the source files that contain your data. The end result is typically an audio file written to your hard drive, but your instruments might be designed with parameters that require real-time MIDI input. You can then play these instruments as you would any software synthesizer.

The real advantage to using this type of environment for synthesizing sound is the enormous power that both programs provide. There's no real limit to the type or complexity of patches that you can build; both languages offer significantly more building blocks than any of the programs reviewed in this roundup. Of course, as with any synthesizer, the more complex the sound, the less polyphony you're likely to get. And though many new "helper applications" are available (for Csound in particular) to make coding your sounds easier, not everyone will find composing with a word processor to be suited to their taste.

If you've never heard the music that sound-programming languages can create, you're in for a treat. Download Csound or the demo version of SuperCollider, and check out the wide range of styles in the examples included with both.

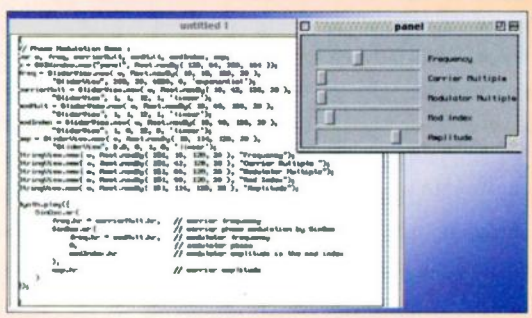

FIG. A: SuperCollider allows the user to build graphic control panels that provide parameter data in real time. Pictured here is the interface to control a phase-modulation patch, along with the code that was used to create it.

the software offers numerous modulation and sync sources. Also available is a Random tuning parameter, which you can use as a modulation destination if you want your sounds to have the classic analog "detuned" quality.

The VCF section contains an assortment of 1-, 2-, and 4-pole filters—with and without resonance—along with the traditional state-variable group (lowpass, highpass, bandpass, and the like) and threshold, comb, and slope. The filters can be configured to run in series or in parallel. An Overdrive parameter is offered for those whose tastes run to the truly gritty, and each filter's cutoff frequency can be modulated, giving you even more possibilities for nonlinear distortion. Two 5-stage envelopes and a dozen internal effects round out the basic toolkit.

Retro AS-1 wins the prize for most presets with more than 1,300, grouped in categories such as FX, Techno, Strings, and Drums. They range in sound from the incredibly fat and juicy "Alaska," to the punchy "Rezo Kizz Bass," to the delicate "Moonflower." I only wish that I could audition them directly from the File dialog box so that I wouldn't have to load them to try them out. In any event, the software clearly provides a huge range of sonic possibilities. It's also well acquainted with all the standard Mac audio drivers, which means it should integrate nicely into any studio environment. And if you happen to have a G4, you'll find that *Retro AS-1* is highly optimized for performance on that system.

***SynC Modular* 1.7 (Win; $49).** Dr. Sync's *SynC Modular* has an interface that resembles *Reaktor*'s, and it uses similar terms for many of its design components. The program groups its modules into several categories, all of which can be accessed from a drop-down menu by clicking the right mouse button. In the menu you'll find Basic modules, which are *SynC Modular*'s most fundamental building blocks; Standard modules, which are the next level of architecture and typically contain many Basic modules as their source; and Library modules, which include macros contributed by users in addition to those provided by the manufacturer. You can create your own macros by loading a Macro module and then building whatever structure you want it to have.

Each of the modules has a host of parameters that you can access and edit by right-clicking on the module's icon. In addition to its sound-making components, you can customize many other aspects of a module—for example, whether it appears in the Panel display, what caption appears on its icon, and, in some cases, what bitmap that icon uses.

A number of shortcuts make revising your designs easy. For example, when you're working in the performance-level Panel display, you can click on any icon and have *SynC Modular* jump directly to the basic component that the icon is controlling, even if it's many layers deep in the synth's structure. The program's ability to render MIDI files is especially useful, and its support for DirectSound ensures that you'll get the most mileage out of your audio hardware. It can also be used as a VST Instrument.

SynC Modular's documentation is available only in a Help file, but it's surprisingly thorough. The documentation includes numerous optimization tips, which cover both specific design issues and global computing considerations. The program's design options are very deep and allow you to work at an incredibly low level of architecture (though you may need to sharpen your DSP chops if you really want to get your hands dirty). Moreover, the program performed exceptionally well on my test platform—equal to, if not better than, most of the others in this roundup. At $49, *SynC Modular* may be the best bargain in the bunch.

***VAZ Modular* 2.1 (Win; download, $282.27; CD-ROM, $338.69).** Martin Fay's *VAZ Modular* hides much of its enormous power behind a simple interface. Rather than use virtual patch cords like most of the programs in this group, *VAZ* employs drop-down menus for making connections among the various modules in your design (see **Fig. 4**). It also has a nifty feature that can "roll up" an individual module so only the title appears. This allows you to create a clean, lean look for your synths and to keep the interface well organized.

The program offers two playback methods: real-time MIDI control and a built-in sequencer. *VAZ*'s sequencer is a very deep feature that has many layers of operation. It offers 16 patterns, each of which can contain up to 16 steps. Each step of a pattern has a Pitch, Rest (mute), Slide (portamento), Accent, and Double (for doubling the step's length) control. You can gang all of

FIG. 4: Modules in *VAZ Modular* are connected by assigning inputs and outputs from drop-down menus. This keeps the interface clean and uncluttered.

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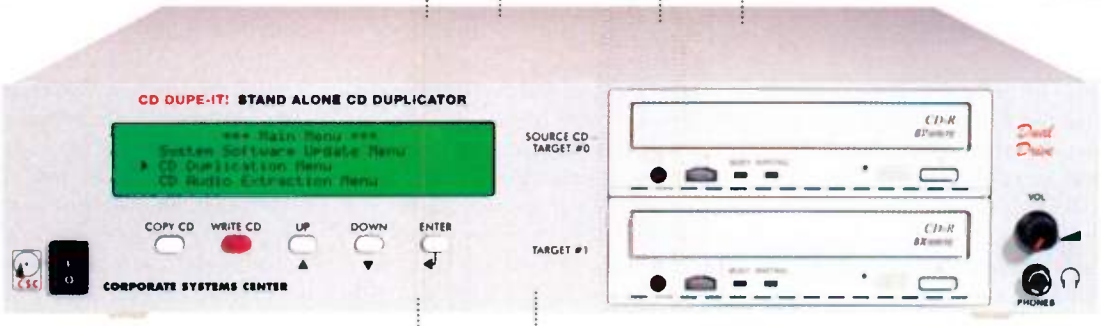

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the controls—for instance, if you want to transpose the pitch of your entire pattern—and adjust the pattern's tempo (from 1 to 255 bpm). But that's just for starters.

The next level of operation controls pattern playback. You can play all patterns in order or set a range (for example, to play back patterns 2 through 6). You can also play patterns in reverse order or use the Song Editor to create a sequence of up to 255 patterns in any order with a unique transposition offset for their pitch. Finally, a pattern randomizer feature allows you to generate patterns randomly and to constrain the options that the randomizer uses. You can limit the range of notes chosen, the number of steps used in each random pattern, whether rests are allowed, and more. If that's still not enough control, you can use a full range of real-time MIDI triggers to determine which patterns will play.

VAZ offers some very powerful sound-design components, including a full complement of audio-generating and audio-processing tools. Among the more unusual are the granular oscillator, which generates a stream of grains at either a fixed or variable frequency; the vowel filter, which morphs between different vocal sounds; and the envelope follower, which converts an audio signal into a control-rate signal. Although its internal effects are fairly limited, *VAZ* is one of the few synths in this roundup that support both DirectX and VST plug-ins. Overall, it's an elegant program that should be high on every musician's list of considerations.

BEAT THE BOX

The stand-alone applications in the next group don't let you design your own synths from scratch; instead, they offer screens full of tweakable parameters that control the generation of sound. Here you'll find familiar sound generators and processors such as LFOs, VCOs, and filters for tweaking your sounds; and for the most part, MIDI control is extensively available. All of these programs model analog synths in one form or another, and several use the "beatbox" model popularized by Propellerheads' seminal *ReBirth* program. Arturia's *Storm* even lets you load several different "analog" tone modules at once and synchronize them to a global clock.

***Vibra9000* (Mac; Studio9000 bundle, $595).** *Vibra9000* is part of Koblo's Studio9000 bundle and is the easy winner of the most-colorful-software award. Its bright green interface features a single resizable screen on which are knobs for controlling all the program's parameters (see **Fig. 5**). You can easily adjust the knobs with the mouse (though you can also use the preassigned MIDI controller for any of them), and a large display indicates the exact value of every parameter you're tweaking. Drop-down menus provide access to other program options.

Vibra9000 groups its functions according to a traditional analog synth design. You can assign any number of waveforms independently to its two VCOs—a knob produces a continuously variable change in the waveform's spectrum as you slowly move it from, say, triangle to sine. Two separate LFOs offer a choice of six waveshapes, and the three ADSR envelopes can be applied to numerous sound parameters. Individual envelope segments can be finely adjusted using 0.001-step increments in a range of 0 to 1, but you can't type in an exact value.

The filter section features 2-, 4-, and 8-pole options along with double, quad, and notch types for added variety. It also offers the unusually named SawComb and Sqr-Comb, which provide multiple resonant peaks that resemble the waveforms suggested by their names. The Distortion parameter is useful for creating sounds that fall into the "nasty and raunchy" category, and the Keytrack setting allows you to determine how closely a filter's cutoff frequency maps to an incoming MIDI note value.

Whipping through the hundreds of presets is made nearly painless with the Page Up/Page Down shortcut (which loads successive patches), and by using the Trigger and Hold controls in the Global screen you can trigger a sound and have it play indefinitely. The Arpeggiator offers several modes for cycling through the preset note patterns.

Vibra9000 provides drivers for nearly every Mac audio protocol there is, but many of the drivers are still in beta. (Check the company's Web site to see if your hardware is supported, so you can avoid using Sound Manager if at all possible.) You can also use *Vibra9000* as a VST Instrument, and it's especially happy when running with Direct-Connect (for use with Digidesign hardware). And though *Vibra9000* is strictly monophonic, its high-quality sound generators and modifiers—coupled with extensive routing options—make it an excellent tool for creating a wide range of analog sounds. It is also a sheer pleasure to tweak. Use it along with the *Stella9000* sampling and *Gamma9000* drum machine modules (also included in the Studio9000 bundle), and you'll have a potent pack of sound sources, all living under the same roof.

***Reality* 1.56 (Win; $379).** Seer Systems' *Reality* was the first professional software synthesizer to hit the market, and though it's showing signs of age, it remains one of the most potent players available. The program's strength lies in the high quality of its physical-modeling technology, which can produce an extensive range of accurate acoustic simulations. You'll find excellent-sounding wind, percussion, and string models, including very realistic flutes, clarinets, marimbas, chimes, and plucked strings. *Reality*'s

FIG. 5: *Vibra9000*'s bright green interface offers a slew of knobs and buttons to twist and tweak. A large display shows the value of every parameter that you modify, and other visual feedback also offers helpful information.

alter ego adds an entirely different set of features to the program: it can also load, manipulate, and play back samples in both WAV and SoundFont formats.

In addition to the physical models, *Reality*'s other main sound source is its group of four oscillators, each of which can be assigned one of six waveforms or three types of noise. The Topology parameter allows you to configure the oscillators in an additive network or in several types of FM arrangements. Each oscillator includes a gain and coarse-tuning parameter—both of which can be modulated by a large number of internal sources or MIDI controller data—a random pitch offset, and an adjustable Velocity-response parameter.

Eight filter types are next in line, followed by four LFOs and four multisegment envelopes. No graphic interface is provided with which to draw envelope shapes; instead you pick one of the default shapes and modify the time or level of the segments using numeric values. Like most parameters in the program, the LFOs and envelopes can have up to two modulators of their own, and you can easily configure the modulation routings using the pull-down windows, where all the options are displayed.

Reality has also overcome one of the major problems in using a stand-alone software synthesizer with a digital audio sequencer, namely, that many software synthesizers take control of your sound card and don't allow the playback of audio from any other source. Using the included Reality Wave Driver, you can route audio from your sequencer directly into *Reality*, mix it with *Reality*'s own audio output, and send the combined signal to your sound card.

A recent update to *Reality* added support for a number of new sound cards, enhanced support for SoundFonts, and solved some user-reported installation problems. It also gives you the option of exporting an entire bank of patch names in the instrument-definition formats of many popular sequencers. Though its interface could use a major overhaul, this robust program has much to like, and the many synthesis options, not to mention the excellent physical models, make *Reality* suitable for a huge range of music projects.

***Storm* 1.1 (Win; $225).** Arturia's *Storm* uses an interface model that is different from the programs covered so far. It offers a "rack" into which you load multiple synth and effects modules that are controlled from its elaborate Composition screen (see **Fig. 6**). Analog synths and drum modules are the story here; *Storm* also has several sample players and a "turntable" on which you can "scratch" samples via MIDI control. Hard disk recording of the program's output is managed very nicely and is well integrated with its other features.

A *Storm* rack can contain up to four modules, each of which can be routed through up to three effects units. (The effects are Delay, Chorus, Distortion, Sequence Filter, and Flanger.) All the synth modules include pattern-based sequencers for triggering notes and provide various controls for adjusting their sound parameters. For example, Arsenic is a bass-line synthesizer that displays a 3-octave keyboard on which you create note patterns. The keyboard can be transposed to provide a 12-octave total range, and each of the 16 notes in a pattern can have one of four Velocity values. Arsenic's sound source is a single oscillator (square, triangle, or pulse) that is hardwired to a filter. The synth includes controls for attack, decay, filter cutoff, resonance, and cutoff modulation.

You mix the output of your rack's various synths in the Composition window, where you'll also find the Editing Bar and several other composition tools. The Editing Bar is a visual representation of your entire composition, with small squares representing individual bars of music. Here you make changes to global parameters (such as tempo or volume) or record changes to the parameters of individual synths. A clever feature called Kepler is used to transpose your patterns on the fly or to create a sequence of key changes. (*Storm* can modify the pitch and tempo of audio in its sampling modules as well as the output of its synths.)

You can record the output of one or more sound modules using *Storm*'s Recorder feature and then immediately use the recording in your compositions. And the entire operation can be done while your piece continues to play—an especially nice feature. *Storm*'s sample-playback options are beyond the scope of this article; suffice it to say that there's a great deal of visual feedback, which makes working with multiple samples quick and easy, even in real time.

Storm is clearly best suited for dance, techno, or any genre that relies on

FIG. 6: The main interface in Arturia's *Storm* resembles a rack full of gear, including synths, effects, a mixer, and a sequencer. All of the devices sync to a global clock.

looping patterns. But the real-time pitch-shifting and time-stretching that you can perform on samples adds another aspect to its personality, one that can lead to many other realms of music making. *Storm* can be used as a VST Instrument, and it's one of the few programs in this article that includes a paper manual. Though its initial release has some rough edges, *Storm* is a very capable performer that can bring you hours of enjoyment.

***SimSynth* 2.6 (Win; $35).** Image Line's *SimSynth* is a vintage-analog-synth emulator with a slew of modern features. It has a number of useful compositional tools as well as some graphic elements that make designing patches quick and intuitive.

The VCO section includes three oscillators, two of which can serve as modulators in an FM configuration (see **Fig. 7**). There are five default wave shapes (saw, square, pulse, noise, and sine), but an Adjust knob allows you to create crossfades among the spectra of the first three. If you select sine, the Harmonic Synthesizer becomes accessible; here you can build composite waveforms using any of 16 harmonic partials. Though you can't type in exact values for the program's various parameters, a clear display shows the current value of the parameter you're adjusting.

Every parameter in *SimSynth* can be mapped to a MIDI controller; you just right-click on the knob and assign it a CC number. (You can also change the range of values that the knob transmits.) For those of you who are graphically inclined, user-created envelopes are available to control pitch, filter cutoff, or amplitude. (All three can be used simultaneously.) By zooming the display in or out, you can make very accurate adjustments to individual envelope segments. *SimSynth* also provides a built-in sequencer—complete with a metronome—for recording and playing back MIDI data, and a 2-octave keyboard enables you to trigger notes and create patterns.

FIG. 7: Each synth component is displayed in its own section of the *SimSynth* interface. Envelopes can be drawn at the bottom of the screen, and various file and playback functions are found at the right.

All output from *SimSynth* can be captured to disk, including the sound triggered by any MIDI data that you've recorded. The many excellent presets, which begin to play when they're first loaded, are a great resource for studying good patch design. (You can get

some really sassy bass sounds if you set the filter cutoff to around 10 percent and experiment with the Adjust knob while using a pulse wave.) To help you build your own presets, the program has a Preset Quick Starter feature that includes suggested settings for various types of sounds, such as Electronic and Brassy—very smart. In fact, that's a good overall description of this well-designed application.

***Probe* 1.5 (Win; $199).** A fairly new arrival in the beatbox world is Synoptic's *Probe*. This capable, 16-voice synthesizer offers up its three oscillators, two envelopes, and two filters in a single main work area. However, lurking beneath the surface are some unique tools that raise the level of functionality a good bit. For example, clicking on VCO I, VCO II, or either of the LFOs opens the Additive Synthesis window, which contains sliders for the amplitudes of 20 harmonic partials. By adjusting the sliders, you can create and then save any number of complex waveforms for use in your designs. You can also view the composite waveform as you adjust the levels of each partial.

Another of *Probe*'s unique features is its ability to analyze the first 20 harmonic partials of a sample file on your drive and load the resulting spectrum into an oscillator's wavetable. Once the spectrum is analyzed, you can alter the amplitude of individual harmonics, which gives you a very simple analysis/resynthesis module. You can also morph between two waveforms, and you can even control the morphing rate with an envelope or LFO.

And speaking of control, you can route an audio signal into *Probe* and use it in various ways. You can employ the audio as a sound source in place of a VCO, or you can extract the amplitude envelope of the audio input (your voice, for example) and use it to control a parameter such as the VCF cutoff. That way, the louder you spoke, the higher the filter's cutoff frequency would be.

Probe's sequencer is very powerful and lets you place audio files or synth patches onto any of its 17 tracks. Up to 16 patterns can play back as many as 24 steps each, and any number of synth parameters can be sequenced over time. It can also import a MIDI file and extract information from one or more tracks for use in your own patterns.

Program Change maps can be easily set up to recall any of *Probe*'s presets, and a large number of options are available to configure the program for use with other MIDI gear. Its high level of customization and numerous

PERFORMANCE PRACTICES

Many factors influence the performance of a soft synth on your desktop, but after surveying a number of the companies mentioned here, I can narrow the issue to several key points. First is the type and speed of the processor that you use. Most programs simply can't provide adequate performance on a slower machine, though some companies—Seer Systems, for example—have written their code in Assembly language, which allows them to get a little extra power from a slow computer. Plan on an absolute bare minimum of a Pentium II/200 MHz and a Mac PPC 604e/266 MHz, which are, according to the companies I contacted, the average minimum requirements. Just remember, however, that more is better.

Sound cards are the next major issue. Here, the quality of your card's drivers is key. Keep your eyes on the manufacturer's Web site for driver updates; you'll be surprised at how often they appear, especially for newer cards. Some soft-synth developers have worked closely with sound-card manufacturers to ensure optimal performance between the application and the hardware. Though this is more common with software samplers, it's a good sign when the hardware and software companies have been in touch with one another.

In the Windows world, using DirectSound improves on the performance that you would get from basic MME audio drivers, and all of the Windows synths support this protocol. The Mac has numerous audio protocols, and you'll have to determine which ones your synth and hardware are compatible with. For example, using ASIO can improve performance on any platform; I switched from MME to ASIO for the E-mu APS card on my PC, and latency in one of the synths went from a completely unacceptable 400 ms to 20 ms. But ASIO support in particular must come from both the sound card and the software; you may not have the right combination of the two.

The next issue is RAM. How much is enough? No one can say for sure, but one of the sequencers that I used on the Mac wouldn't even load a synth plug-in until I bumped up its allocation to more than 50 MB. All manufacturers agree that synths are RAM hungry, so like they say at holiday time, give and give generously. On the Mac, just be sure that you leave enough RAM for your computer to manage its other tasks.

There are dozens of other tweaks that you can use to eke out the best performance from a soft synth, such as turning off Virtual Memory on the Mac and loading only those extensions and control panels that you really need during a work session. (Check Extension Manager to find out which extensions are loading, or buy a copy of *Conflict Catcher* for even easier extension control.) PC users should shut down Autonotification in their Device Manager, because it periodically checks to see if a CD has been inserted into your drive, which could happen at just the wrong moment.

Today, having a powerful computer is a prerequisite for top performance; but there are many other considerations, including your audio hardware, the audio protocols that your platform supports, and the options that the software itself provides. No single solution works in all cases, but with a few tweaks (and a little luck), you should be all set.

high-quality audio components make it an excellent choice for your analog needs.

ALTERNATE MODES

Most of the programs in the next category accept MIDI data for the control of various sonic parameters but are not primarily intended for use with a keyboard controller. Instead, knobs, sliders, and other interface elements are available for triggering playback, and you'll also find the occasional automated control to relieve you of some of the real-time duties.

These programs all generate sound in real time using various synthesis methods, some of which are far less common than those used in the programs I've covered so far. The big winner here is granular synthesis, which is at the heart of several members of this group. Though at first you might suspect this category of being on the fringe of the musical landscape, you'll be surprised by the range of sonic options that the following programs provide.

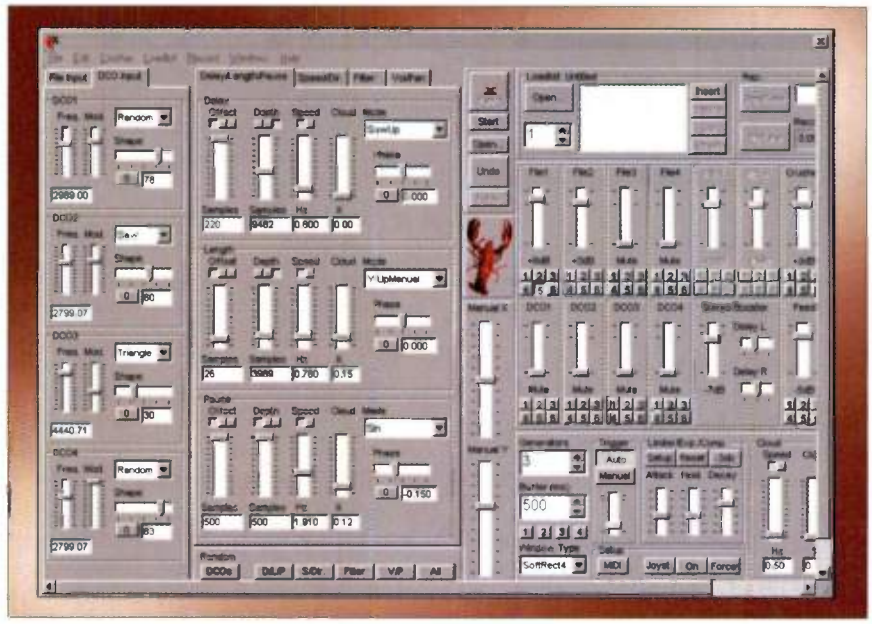

FIG. 8: *CrusherX-Live* offers a huge number of sliders to control the various grain parameters that it uses. The four oscillators at the left of the screen (marked "DCO") are at the heart of its synthesis engine.

***GranuLab 8* (Win; $20).** Although granular synthesis is one of the most efficient ways to generate unique and colorful sonic landscapes, it's a method that hasn't yet appeared in major hardware devices. (Symbolic Sound's Kyma System is the one exception.) However, granular-synthesis software is another story. Rasmus Ekman's *GranuLab 8* is one of several

powerful granular synthesizers, and, like some of the others in this group, it allows you to synthetically generate thousands of small sonic grains or to slice and dice existing sound files into minute fragments.

GranuLab 8's single screen consists of numerous sliders that control the various parameters of each grain. In addition to sliders that set the base value for pitch, grain density, amplitude, stereo position, and glissando rate, there are separate sliders (one for each base slider) that add an amount of randomness to each parameter's base value. (When using samples, you can also assign a value to the sample start position and add a random offset to that point for each file.)

You can run up to eight different sets of controls, each producing a separate grain stream, and you can mute or solo individual streams. Depending on the density of each stream, that may be more than many computers can manage. You can also save 20 snapshots of your settings at any given point and instantly toggle between settings, or you can set a parameter change time to determine how long the transition time from one snapshot to the next will be. (*GranuLab 8* morphs between the two settings over the time that you indicate.)

Additional real-time control is available in the Patch Gesture window. This window, which functions like the Vector control in some Korg devices and the Blue Window in Arboretum's *Hyperprism*, lets you use your mouse to "morph" among four different snapshots. As you move the mouse to extreme corners of the window, the snapshot you've assigned to that corner takes precedence. Finally, you can assign a Program Change to recall snapshots, or you can simply use MIDI controllers to manipulate most parameters.

A freeware version of *GranuLab 8* with only one grain stream is available at the developer's Web site, but if you can see a world in a grain of sound, pick up the registered version and experience the entire universe.

***CrusherX-Live* 1.4 (Win; $39).** Another contender in the granular group is Joerg Stelkens's *crusherX-Live*. This awesome little application has a host of features for granulating samples of real-time audio input or generating grain clouds from scratch. You can even mix multiple sources in the same session. The program's parameters can be controlled using several types of MIDI messages, and it's well suited for use with alternate controllers such as a joystick. Though the interface won't win any prizes for elegance, the sliders and buttons are easy to manipulate, and the program's overall design is very intuitive.

At the heart of *crusherX-Live* is a set of oscillators that are summed and sent to the Crusher panel (see **Fig. 8**). The panel uses the oscillators for input and offers numerous controls for their manipulation. Like many implementations of granular synthesis, *crusherX-Live* allows you to control grain frequency, amplitude, timbre, length, density, and time between grains. Random offsets can be added to any parameter using several types of random functions; and you can generate multiple grain streams simultaneously. Several windows provide a graphic display of the state of the various parameters.

CrusherX-Live excels at real-time performance and is a powerful instrument once you master its intricacies. For example, the Loadlist offers the functionality of a playlist; you can use it to load numerous preset parameter settings and switch among them—manually or using MIDI events—as the program plays back. With a set of buttons at the bottom of the screen you can randomly assign a new set of values for one or more parameters, and the unlimited Undo/Redo (which can be enabled via MIDI notes or with a button on a joystick) lets you return to any point in your session.

The program's ability to interact with a force-feedback device (whereby the value of different parameters causes you to feel greater or lesser pressure at the controlling device) is yet another unique feature

Soft Sites

You can find additional information about the synths covered here at the manufacturers' Web sites. Most of these programs have demos available and nearly all can be purchased online.

Product Name	Manufacturer	Web Site
Buzz	Oskari Tammelin	www.buzz2.com
Chaosynth	Nyr Sound	www.nyrsound.com
CrusherX-Live	Joerg Stelkens	www.stelkens.de/sk/crusherx/index.html
Csound	Barry Vercoe	www.csound.org
ES1	Emagic	www.emagic.de
GakStoar delta	Lin Plug	www.linplug.de
GranuLab 8	Rasmus Ekman	hem.passagen.se/rasmuse/Granny.htm
Model-E	Steinberg	www.us.steinberg.net
Probe	Synoptic	www.synoptic.net
Pro-Five	Steinberg/Native Instruments	www.us.steinberg.net
Reaktor	Native Instruments	www.native-instruments.com
Reality	Seer Systems	www.seersystems.com
Retro AS-1	BitHeadz	www.bitheadz.com
SimSynth	Image Line	ellisdee.onestop.net/SimSynth2.htm
Storm	Arturia	www.arturia.com/en
SuperCollider	James McCartney	www.audiosynth.com
SynC Modular	Dr. Sync	www.mtu-net.ru/syncmodular
Tassman	Applied-Acoustics	www.applied-acoustics.com
VAZ Modular	Martin Fay	www.software-technology.com
Vibra9000 (part of the Studio9000 bundle)	Koblo	www.digidesign.com/prod/koblo
WaveWarp	Sounds Logical	www.soundslogical.com

that encourages live experimentation. *CrusherX-Live* can produce an enormous range of interesting textures, and one of them might be just right for your next gig.

***Chaosynth* 1.0 (Mac/Win; $40).** Nyr Sound's *Chaosynth* offers one of the most unusual means of sound generation that you're likely to find. The program combines granular synthesis with an algorithmic process known as *cellular automata* and allows the user to "perform" many of the synthesis parameters in real time. The mouse is the primary control tool, but MIDI controllers and even Note On messages can be used to alter many parameters as the synth plays.

Cellular automata is a process that employs a set of rules to determine the way in which a series of values evolves over time. As implemented in *Chaosynth*, the process makes determinations about what notes will be heard at a given moment and governs the transition from one set of parameter values to the next. The result can be anything from massively dense grain clouds to subtly evolving, almost organic textures that swirl throughout the spectrum.

User interaction occurs in many ways. For example, you can drag the mouse in the Frequency panel to change the number and range of frequencies that are included in the grain stream (see **Fig. 9**). Or you can draw curves in the Oscillator panel to alter the amplitudes of the various frequencies. Grain size is controlled in its own dedicated window, and among the many options are controls to alter the behavior of the automata process and to apply reverb, filters, and envelopes.

One of my favorite features is the ability to convert grain events into MIDI notes, which means you could have one layer of sound that uses the basic waveforms in *Chaosynth* while another layer plays a MIDI-orchestrated version of the same data. The possibilities are endless.

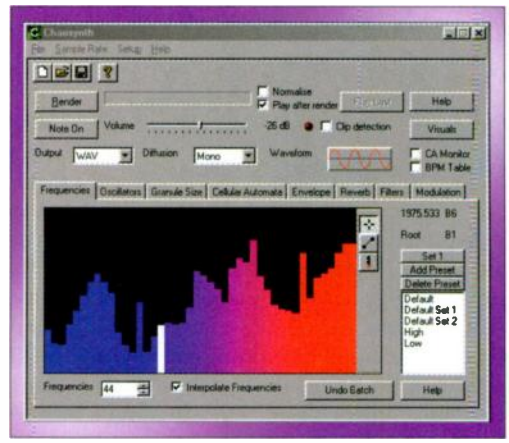

FIG. 9: User input to *Chaosynth* comes in many forms. By sweeping the mouse across the Frequency panel, for example, you can "perform" the frequencies that are used by each oscillator.

Chaosynth includes an excellent tutorial on the synthesis and algorithmic methods it uses and provides numerous interesting presets to get you going. If you're looking to add an entirely new dimension to your audio toolkit, *Chaosynth* is definitely worth checking out.

***WaveWarp* 1.2 (Win; $199).** *WaveWarp*, from Sounds Logical, is a massive construction kit for building synth

networks and signal-processing routines; and though it doesn't yet accept MIDI input, it offers numerous ways to control the generation of sound in real time. For example, you can easily build complex additive-synthesis networks that have controls for each partial's amplitude or create subtractive synthesizers using any of the program's various high-quality filters. You can also make "self-running" FM designs that generate hours of slowly evolving ambient textures.

WaveWarp's modules are accessed through tabbed folders that appear at the top of the main screen. You build a synth network by dragging modules onto the Drawing Board, where you make the connections and set the parameters for each component in your design. Clicking on any component brings up a window where its parameters can be controlled as the sound plays.

The list of sound-generating modules is exhaustive. In the Signal Generating category, you'll find sweep generators; sine, square, and triangle wave generators; chaotically controlled signals; and more. Reverbs, pitch-shifters, and spectral transformers are also provided in large quantity, and delays, choruses, and flangers are just a few of the many additional tools. Don't expect any automated patch-generation options here, though; *WaveWarp* is a serious sound programmer's toolbox, and, as such, it demands a fair amount of knowledge on the subject.

Using *WaveWarp* as a stand-alone synth would keep you busy for some time, but the new 2.0 version, which should be shipping by the time you read this, allows you to use your creations as DirectX plug-ins. Though it isn't currently suitable for live performance using a MIDI controller, it has an enormous range of powerful sound-producing modules that you can combine in an endless number of configurations. If you're a hacker at heart, *WaveWarp* should be in your workshop.

PLUG-IN SYNTHS

Running a software synthesizer from directly within a host application is now quite popular; Steinberg deserves the lion's share of the credit for making its VST Instrument format the method of choice for nearly all software of this type. Dozens of VST-format synths run as plug-ins, and using them can be as simple as assigning the output of a sequencer track to the VST Instrument and pressing the Play button. (Different hosts—for example *Cubase VST* and *Logic Audio*—use different means of employing VST Instruments.)

VST Instruments offer several special advantages. For example, you don't need to use virtual MIDI drivers or IAC connections because the Instruments receive their control data directly from sequencer tracks. You can also automate all of your synth patches' parameters using MIDI controllers, and you can usually save complex synth designs right along with your sequencer files. Furthermore, you don't have to worry about which internal effects a VST Instrument offers, because you can route its audio output through any VST or DirectX effects plug-in on your system.

You should make sure that you have current, stable audio drivers installed to get the best performance from a VST Instrument; and remember that you can't tweak the latency of a VST Instrument separately from that of the other audio sources you're using in your digital audio sequencer. As stated earlier, don't expect to stay in sync with the music while performing a VST Instrument live from an external keyboard and while multiple tracks of audio are playing back (much less multiple effects). Most systems simply can't offer that level of performance yet.

With the exception of Emagic's *ES1*, all of the synths in the next group are intended for use as VST Instruments. Though VST Instruments come in several varieties, most model classic analog synthesizers. (An excellent resource is Ben Turl's K-v-R VST Instrument Banks site at www.k-v-r.freeserve.co.uk. Here you'll find the occasional virtual bass or drum machine as well as early announcements of new synth plug-ins.) Keep in mind that many of the synths I've covered so far, such as Native Instruments' *Reaktor*, can also be used as VST Instruments. That gives you even more options to use in this format.

If you're a PC user and you don't have a VST 2.0 compatible application, you'll be pleased that *VST Adapter* from FXpansion (www.fxpansion.com) lets you load a VST Instrument into many DirectX-compliant programs. (Problems have been detected with certain DirectX hosts. Ask your software manufacturer about any known incompatibilities.) Regardless of the host program, you'll find that these plug-in synths offer considerable editing flexibility and integrate extremely well with your sequencer environment.

***Model-E* (Mac/Win; $199).** Everybody knows that analog modeling has been a hot ticket for the past few years, but no one could have suspected that so many legacy hardware devices would be modeled directly in software. Among the plug-ins that model vintage hardware is Steinberg's *Model-E*, which has the look and feel of a Minimoog Model D (see **Fig. 10**). No doubt

FIG. 10: Several new VST Instruments are modeled after analog synth hardware. Steinberg's *Model-E*, pictured here, is the spitting image of the Minimoog Model D.

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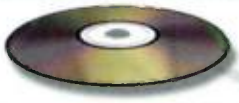

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Model-E offers enhancements that include expanded polyphony, complete MIDI control, and 16-part multitimbral capability. In place of the original Moog's keyboard, you'll find a MIDI setup area where you pick programs, banks, and MIDI channels, and assign the amount of polyphony that you want to dedicate to the synth. (The maximum is 64 voices.) You can send notes directly from an external keyboard, but most likely you'll trigger the synth from a track in your host sequencer, just as you would with any of the VST Instruments discussed here.

Each of *Model-E*'s three oscillators offers independent level control and a choice of six waveforms with a frequency range that spans 6 octaves. Oscillator 3 can be used as a mod source, such as an LFO. A noise generator provides an additional sound source and is especially useful for percussive timbres.

The VCF is switchable between 2- and 4-pole and can track the pitch or Velocity of incoming MIDI notes. The filter also has a dedicated envelope generator and can be modulated by oscillator 3 or from your controller's mod wheel. Glide and Tuning settings are found in the Controller section, where you also enable the different modulation options.

Certain trade-offs are always required when modeling hardware devices in software, but I still feel that the control of *Model-E*'s rotary knobs might be improved. Unless you click on the very tip of a knob when you adjust it, the knob jumps well beyond its current setting, making it difficult to fine-tune a parameter. One good solution is to use a physical MIDI controller to make adjustments, which also allows you to tweak several controls at once.

Regardless of the minor problems, *Model-E* is a massive achievement in modeling technology and brings the sound of one of the most popular synths of all time back to the studio. It should conjure up fond memories for quite a few people.

***Pro-Five* (Mac/Win; $199).** *Pro-Five* from Steinberg/Native Instruments is a virtual model of Sequential Circuits' Prophet-5; and like *Model-E*, it adds significant new features to the original (see **Fig. 11**). *Pro-Five* tops off at 32 voices, which should be well beyond the ability of most host systems. (If you need additional voices, you can load multiple instances of the instrument.) For ultra-fat sounds, you can put the device into Unison mode, in which all available voices play a slightly detuned version of the same note.

Like the Prophet-5, *Pro-Five* provides two oscillators per voice, each of which can be tuned over a 4-octave range in semitone increments. Oscillator A offers sawtooth, pulse, or a combination of the two; oscillator B adds a triangle option. (If you select the pulse wave, you can vary its duty cycle from 1 to 99 percent.) Oscillator A can be hard-synched to oscillator B, which also doubles as an LFO. As with *Model-E*, you can add a noise generator to the mix.

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Triggering notes on *Pro-Five* is especially easy, as the plug-in sports a built-in retractable keyboard complete with pitch-bend and mod wheels. Though the knobs are fairly small, moving them with the mouse is easy; just grab a knob and drag the mouse up or down. (A rotary-movement mode is also available). Numerous modulation sources—mostly controllable from the Poly-Mod, LFO, and Wheel-Mod sections—provide the full range of sound-design options that polysynth users have long enjoyed.

Pro-Five ships with 512 sounds that are organized by Program, Bank, and File. But unlike *Model-E, Pro-Five* doesn't display the name of a patch when that patch is loaded. The copy-protection scheme gives you the option of installing a 50 MB file onto your computer or keeping the CD-ROM in your drive when you use the plug-in.

Native Instruments prides itself on the accuracy of *Pro-Five*'s emulation, "right down to the waveform used in the test tone." It's been many years since I had my hands on a Prophet-5, but from all recollections, this year's model has got it spot on.

FIG. 11: *Pro-Five* is a virtual version of Sequential Circuits' Prophet-5 with a number of digital enhancements. It offers up to 32-note polyphony and has a keyboard that can be toggled on and off.

***GakStoar delta* 1.1 (Win; $75).** Lin Plug's *GakStoar delta* is a virtual analog synth that offers 13 waveforms for each of its four oscillators. The frequency of each oscillator is tied to the same envelope, but each oscillator can have its own unique amplitude envelope. Oscillators 1 and 3 can be used as modulators for oscillators 2 and 4, giving you some very flexible AM and FM options. Unlike many analog synths, *GakStoar*'s oscillator output can be sent directly to the main outs without passing through the customary filter sections.

The filters can run in parallel or in series, and each has settings for type (highpass, lowpass, and bandpass configured at 6, 12, 24, and 36 dB per octave), cutoff, resonance, keyboard tracking, and routing.

A large panel in the center of the interface provides easy routing of up to 17 modulation sources to 17 destinations. You can also scale the modulation amount in this window. Four LFOs, each with a choice of seven waveshapes, can be applied to a destination. Attack and Decay times for the LFOs can be adjusted from 0 milliseconds to 10 seconds in 1 ms increments, but it's not really possible to employ every conceivable value within that range if you're using the mouse to adjust the knob. The LFOs can be synched to the tempo of your sequence.

GakStoar has a single Undo level, but even that is more than many of the other programs in this roundup. Its Copy function makes it easy to copy the settings of one oscillator or envelope and apply them to another. A very thorough manual explains all of the program's features and adds to the appeal of this attractive synth.

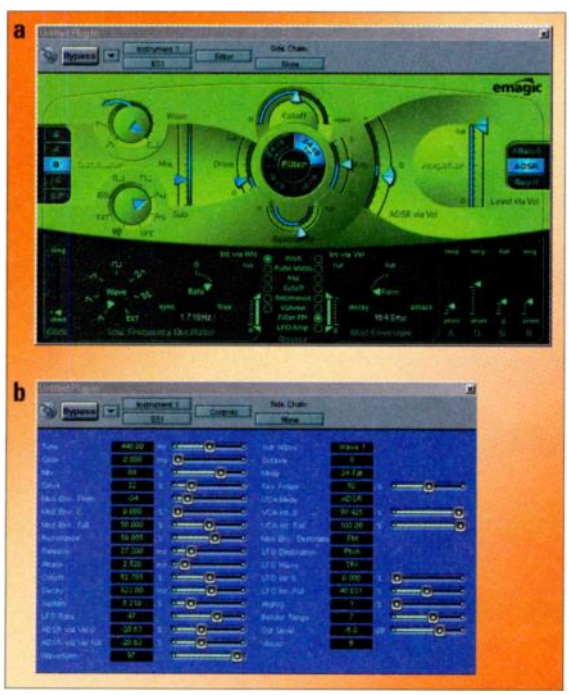


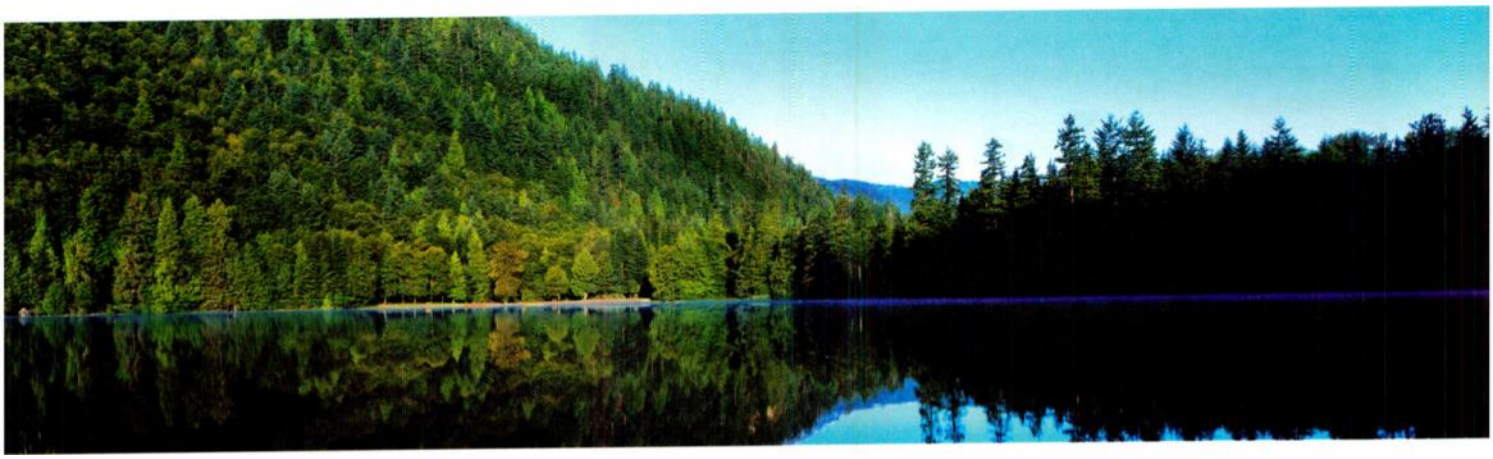
FIG. 12: Emagic's *ES1*, which runs inside the *Logic Audio* series of sequencers, can be operated in two modes. The graphic synth view with knobs and sliders (a) is suitable for real-time control. The slider interface (b) is better suited to offline tweaking.

***ES1* (Mac/Win; $99).** Emagic has released a plug-in for its *Logic Audio* series of sequencers that has capabilities equal to the best of this bunch. *ES1* is a 16-voice, single-oscillator synth with the unique ability to toggle between a graphic synth-panel view and a simple numeric display containing all of the synth's parameters (see **Fig. 12**). This dual operating mode allows you to choose between a user interface optimized for live performance and one best suited to offline tweaking. Up to eight instances of *ES1* can be run in parallel, provided you have the computing resources to tackle that task.

The single oscillator spans a 5-octave range and offers triangle, sawtooth, and variable-pulse waveforms. A suboscillator, which can be toggled on and off, tracks the pitch of the main oscillator and can be assigned its own waveform or noise. The LFO offers six fixed waveshapes, including a slowly evolving random waveform. The LFO can also sync to the tempo of your MIDI tracks; you can choose a number of divisions and a multiple of the tempo, or it can be freely set within a range of 0 to 24.5 Hz.

ES1's filter can be switched to four different rolloff settings: 12 dB, 24 dB, 24 dB Classic, and 24 dB Fat. (The Fat setting boosts the low end that's lost when resonance is used; Classic does not perform that compensation.) Filter parameters include Cut-Off, Drive (overdrive), Resonance, and Key (keyboard sensitivity), all of which can be modulated. You can also route an audio track through *ES1*'s filter, a feature provided by a number of early synths, including the Minimoog Model D. Because of *ES1*'s integration within *Logic Audio*, you

From a land of forests & fables



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can automate most other parameters, and *Logic*'s own effects plug-ins can be applied to *ES1*'s output.

A single envelope generator (ADSR) can be assigned to most parameters, though the routing method isn't as intuitive as it could be. You can also use the AGateR setting to deactivate the Decay and Sustain segments, or you can enable the GateR option to force the Attack time to 0 seconds. That leaves only the Release segment in effect.

ES1 ships with a number of presets, many of which emphasize the lower depths of the audio spectrum. "FM Bass," for example, has a very rich low end, and "Dry Bass," "DB Bass," and "Beyond Bass" also move an enormous amount of air. (Try opening the filter on some of the bass presets to give them a nasty bite.) The snare, hi-hat, and toms aren't quite up to the standards of some of the other drum presets in this roundup, but you won't limit yourself to presets when you have this much horsepower on hand.

RELEASE SEGMENT

Desktop musicians today have an awesome amount of computing power available to them, and software synthesizers offer a great way to use that power. Unlike most of your synth hardware, soft synths improve with age as you put newer and faster processors at their disposal.

I hope you're inspired to add one or more of these programs to your audio toolkit, and I'm sure you'll find that many of them can add considerable new sonic possibilities to your rig. (To purchase or learn more about the products in this article, see the table "Soft Sites.") Soft synths are one of the most exciting new categories of music software to come along in recent memory, and you can bet they'll remain a major focus for creative manufacturers and developers for years to come.

EM *associate editor* **Dennis Miller** *now has more software on his computer than at any time in history.*

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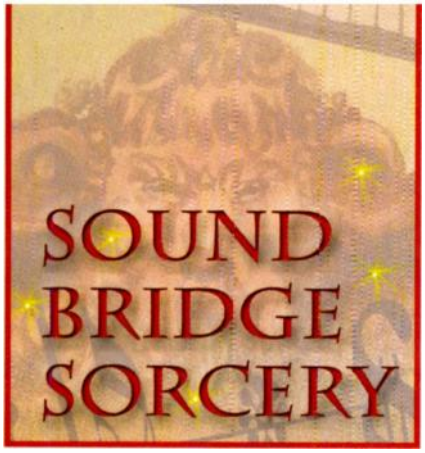
Imagine that your Alesis synthesizer could suddenly exhibit some of the power of a 16 MB sampler and a 100-song sequencer. As long as your imagination is running wild, let's add up to 256 extra Program slots and 200 multitimbral setups. You don't need to fantasize—it's all there waiting for you, thanks to Alesis's in-house software program, *Sound Bridge*. This "Master Class" will walk you through the basic steps needed to make these dreams a reality. Let the burning begin!

CARD-CARRYING MEMBERS

People will often choose one product over another based on features that they may never use. Sometimes it's just nice to know that if you ever need a particular feature, it's available to you. Most Alesis synthesizers and the latest Alesis drum module offer you such rarely tapped capabilities. For example, on the backs of these units are slots for cards that provide space for many additional samples and programs. You can even store MIDI sequences that are instantly available for playback.

Even though many Alesis owners have taken advantage of the wide variety of ready-made sound cards (known as QCards), few have dared to explore the many exciting possibilities of rolling your own. Wouldn't it be amazing to have your background vocal hits ready to beef up your live performances? How about having all of your best groove loops ready to go with zero loading time? Wouldn't it be convenient if you could have up to 100 of your own sequences ready to play without a computer or hardware sequencer? In this "Master Class" I will cover the tools, software, and software that you'll need to create your own sound cards.

BY DANIEL FISHER



Bridging the Gap

When the QuadraSynth line came on the market back in the early '90s, it offered a lot of high-powered features. Included in these features was a card slot for additional samples and programs. To keep the price attractive, Alesis left off certain features such as onboard sample RAM, user sampling, and an onboard sequencer.

Every synthesizer manufacturer has its own software and hardware development tools: the engineers at Alesis had a handy little tool called *Sound Bridge*. That software let them gather their samples and programs and burn them to a card. Why doesn't the public have access to these proprietary tools? Development software tools are often very quirky and nowhere near user-friendly. These tools are often cobbled together as each engineering need pops up, which hardly makes for a consistent and streamlined user interface. If you add the manual that would have to be written and technical support issues to this user-hostile mix, you would probably agree that such software is best left to the engineers. Despite these very daunting considerations Alesis released *Sound Bridge* software as a freeware tool for QuadraSynth owners. Version 3.0 is the most recently available version as of this writing.

FIG. 1: The Card Info window lets the user customize the name of the card and assign a version number. Entering the size of the card will allow you to keep tabs on remaining memory.

To the Bridge!

As it is when you explore any new computer-based tools, there is a learning curve and an initial frustration hump that you have to get past before you can reap the benefits of your labor. As a programmer, I've made frequent contributions to Alesis's factory sounds. As a QS owner, I decided that it was crazy not to use all the power that *Sound Bridge* gave me. I decided to dig in, learn *Sound Bridge*, and share my findings. Along the way I've discovered a few helpful hints that should provide you with snag-free satisfaction. Although this "Master Class" will help you move along quickly and painlessly, it is not meant to be a substitute for reading the well-written *Sound Bridge* manual that comes with the software. Let's get started.

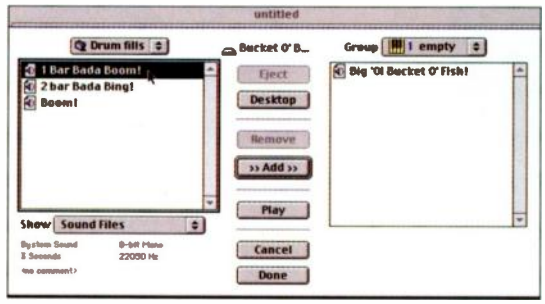


FIG. 2: In the Add File window, I've selected a single sample from the onomatopoetically titled list at the left. Note that I can assign the samples to any group that I choose by using the Group pull-down menu on the upper right.

Let's Connect

Check the *Sound Bridge* manual for your specific computer platform and interface (MIDI or serial connector), and follow the instructions regarding necessary drivers and MIDI Setups. Note that *Sound Bridge* expects to see responses from a single QS or DMPro device. Temporarily remove or power down any other Alesis synths in your MIDI system to prevent problems.

Creating a Project

A *Project* is any collection of instruments, samples, programs, and sequences that you choose to send to your sound card using *Sound Bridge*. Once this data is in place, you can use *Sound Bridge* to compile it onto a PC Card or store the Project on your computer's hard disk for later use. Place your PC Card securely into the top card slot (if you have two slots) found in the back of your QS or the front of your DMPro.

Open the installed version of *Sound Bridge* on your computer. This will automatically begin a new Project and open the main window. Go to the Project menu and select Card Info. In this window (see **Fig. 1**) you can input your own card name, your personal version code, and the memory size of your PC Card. Once you've indicated the memory size you'll be able keep track of the remaining sample memory in the bottom of the main window. You will also need to indicate the type of instrument this card is intended for (QS synth or DMPro).

Group Huddle

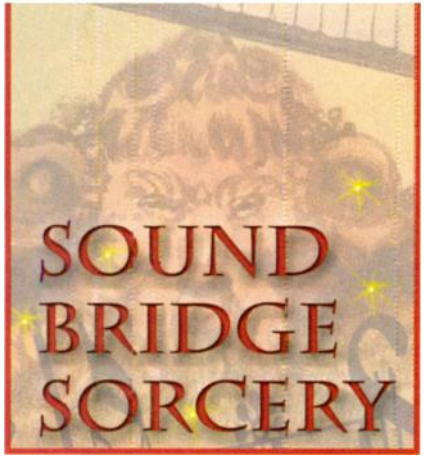
On the left side of the Project window you'll see 13 groups: 8 Keyboard, 3 Drum groups, a Sequences group, and a group for Programs. These are for the different types of data that you may wish to send to your sound card.

You can load Instrument and Sound files into either the Keyboard groups or the Drum groups. Standard MIDI files are loaded into the Sequences group, and SysEx files of Program and Setup banks are loaded into the Programs group. For more info on which specific types of files are supported for each Alesis device, see the sidebar "*Sound Bridge* Compatibility."

Loop and Load

For our first experiment let's try something simple—a single sound effect or rhythmic loop will do. Click on the Add Files icon (the folder with the red arrow). On the left side you'll see a pull-down file menu that will allow you to navigate through your hard drive or CD-ROM until you find the desired samples. Compatible samples will be listed in the window on the left. By double-clicking on a sample, you will transfer it to the right window (see **Fig. 2**). You have now placed the sample into group 1. If you wish to place it in a different group, use the pull-down menu on the right side to select a different group.

For manufacturer information, please see Contact Sheet, p. 205.



For now, double-click on a single sound effect or rhythmic loop to make it part of group 1. Next, press the Done button. After a moment you'll be back to the main window. If you click on the group 1 icon you'll see your sample listed. Double-clicking on its icon will open the Voice Info window where you can rename the sample. Don't worry about the Lower Key and Upper Key parameters—they will be grayed out at this time. If you double-click the sample in the Voice Info window you can indicate the sample's Root Key. The default is Middle C (or C3). Checking the Smooth Attacks box will remove an unwanted click from the sample start if it begins at a nonzero value.

The next three choices—Use Left Channel Audio, Use 50/50 Mixed Audio, and Use Right Channel Audio—allow you to make use of stereo samples. By adding a stereo sample twice to the Group, you can then select Left Channel for one and Right Channel for the other. You can have one layer use the Left Sample and another layer use the Right Sample when you use the samples in a program. By panning these layers in opposite directions you will then hear the original stereo image. An easier way to deal with many stereo samples is to go to Preferences in the Edit menu and check Auto-split Stereo Files. This will automatically split stereo samples into separate mono Left and Right samples for you. When you're finished, press OK to return to the main Voice Info window, then press OK once more to return to the main Project window.

Wide Receiver

Now that you have a simple project with a single sample in group 1, it's time to send the sample to your QS or DMPro. Simply click on the rightmost icon that has a picture of a red arrow pointing from a computer to a synth. The display of your QS or DMPro should say Receiving MIDI. That's all there is to it! It might take a little while for the data to reach your synth (you'll definitely want to grab a cup of coffee for larger dumps). Now, let's set up your synth so that you can play your new sample.

Trick of the Trade

There's a neat little hidden trick to make an initialized Program. Simply power cycle the QS while pressing the 0 and 3 buttons—you'll then have a blank Mix and a blank Program. The unit will default to the Mix mode on power-up, so you'll have to press the Program button one time to see the blank Program. Save this program to a location that you don't mind overwriting.

Next, press Edit Select and then the Sound 1 button (on QS6 and later synths, press the 00 Piano button instead). Using the Page Right button (to the left of the screen), press until you see the Group parameter. Now use the Value Up button or the Value Slider to go all the way up to your new group. Now try any key on your synth. Voilà! There's your sample, and it will play across the entire keyboard.

WHAT YOU'LL NEED

Alesis instrument:
Any Alesis QS synth from QuadraSynth+Piano to current QS series or DMPro drum module.

PC Card:
Any Type I SRAM PC Card, all AMD "C" series and "D" series 5V-only Type I Flash cards, and AMP Flash 5-C series 5V-only Type I Flash cards. These PC Cards may be of any memory size, but user sample storage will be limited to a maximum of 8 MB.

Macintosh computer:
Macintosh 68030; 8 MB RAM; Mac OS 7.1; and any standard Mac MIDI interface or direct QS/DMPro serial connection.

PC computer:
Any PC running Windows 3.1 or Windows 95, 98, or NT; 8 MB RAM; and any standard PC MIDI interface or direct QS/DMPro serial connection.

Cables:
2 MIDI cables (between MIDI interface and QS) or a single serial cable (between computer and QS).

Software:
Sound Bridge for burning samples, programs, and sequences to the PC Card; and *FreeLoader* (or any SysEx compatible software sequencer) for receiving and sending program data.

Optional equipment and software:
A CD-ROM drive for loading SampleCell files, sample files, audio CD samples, and other data.

Digidesign's SampleCell to audition and create SampleCell instrument files. (Note that *Sound Bridge* does not require that you own SampleCell hardware in order to use SampleCell instrument and sample files. These may be loaded directly into *Sound Bridge* from CD-ROM, floppy disk, or hard disk and then transferred directly to the PC Card.)

Samples/Instrument files:
Sound Bridge instrument files; SampleCell I, II, and SampleCellPC instrument files; AIFF sample files; *Sound Designer* 1 and 2 sample files; Mac OS system sounds; and Windows WAV sample files. (Other file types are possible, provided that you input their correct parameters in the Voice Info menu.)

Sequences:
Any Standard MIDI File, Type 0 or Type 1.

Program banks:
Any QuadraSynth/QS and DMPro raw SysEx bulk dumps.

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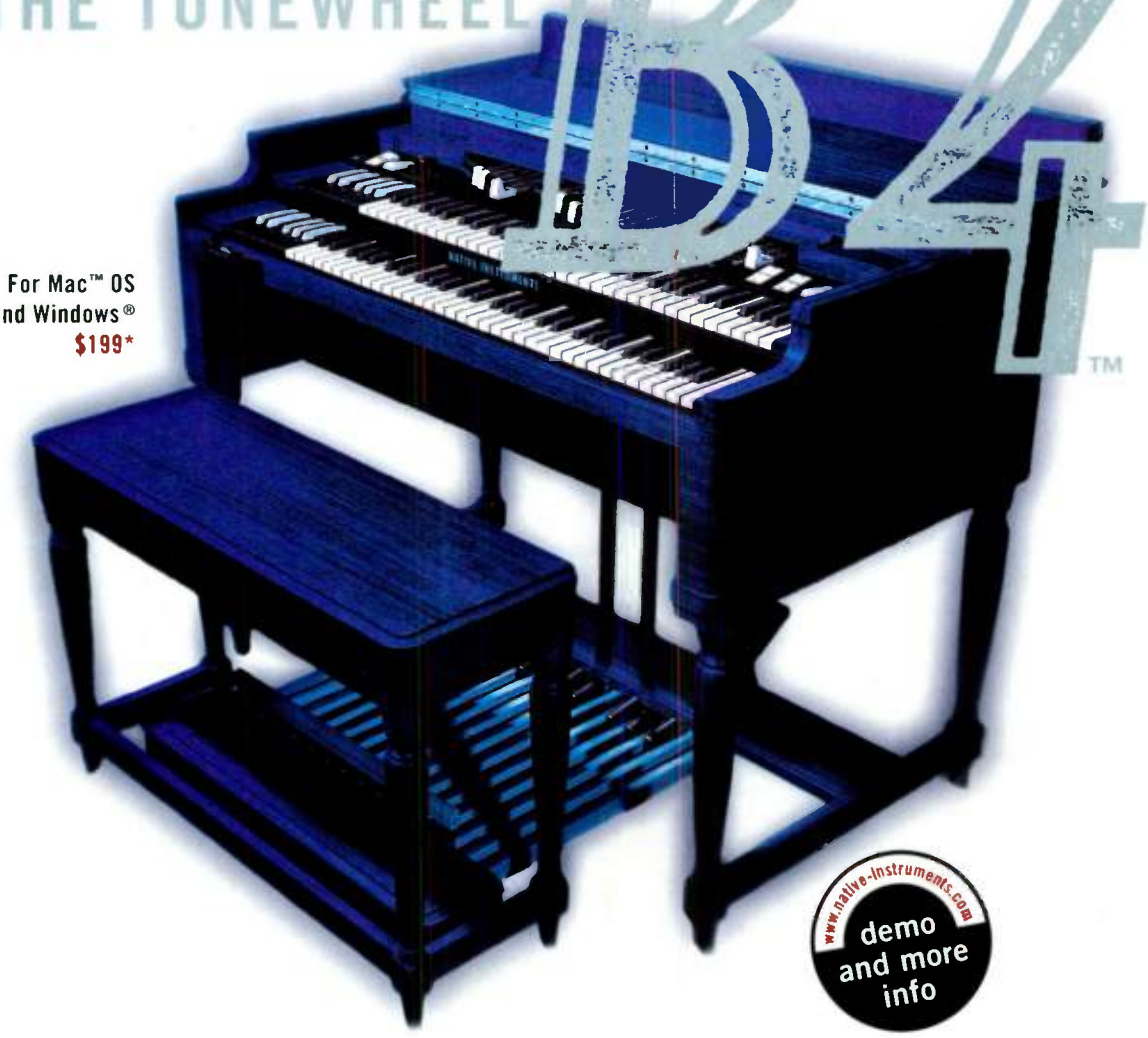

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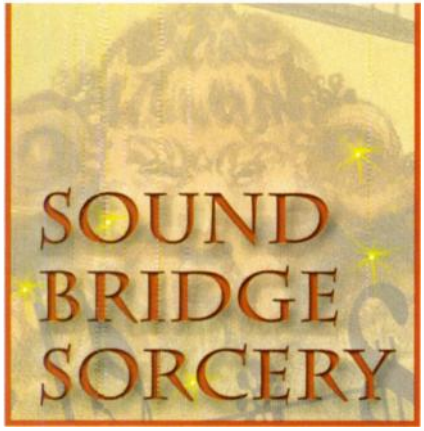


Cellular Structure

Now that you've got the hang of sending a single sample to your sound card, it's time to experiment with the next level of complexity. The easiest way to get multisampled instruments into your sound card is to import them from a SampleCell CD-ROM. Go back to your Project (or start a new Project if you wish) and click on the leftmost icon with the picture of the keyboard and light bulb. (This feature is available only with *Sound Bridge* 3.0 and higher.) That will open the Create Instrument window. On the file pull-down menu you will be able to go to your SampleCell Instruments. When you see them in the left window you can double-click on them to put them into the right window. Once again, you can use the right pull-down menu to choose which group will contain each instrument.

At this point you can press Done and then press the Send to synthesizer icon. It will probably take longer than your previous sample dump to get all that data across. If you really want to speed up the transfer time, go to the Alesis Web site and download the most current serial driver for your computer. By using a serial cable directly to your computer instead of the MIDI interface, you can get up to a 4× increase in transfer speed. Be sure to follow the directions in the *Sound Bridge* manual.

Sax, Violins, and Hair Loss

The most exciting aspect of *Sound Bridge* 3.0 is that you can now create your own sample keymaps; you no longer need to own a SampleCell editor. Simply click on the leftmost Create Instrument icon in the Project page and use the left pull-down menu to get to the source of your multiple samples.

Here are several tips that will save you days of exasperation and hair pulling—pay close attention! Make sure that the instrument samples you have are already in perfect tune: there are no provisions in a *Sound Bridge* Instrument to fine-tune each sample. Next, make sure that the name of each sample contains its root key—for example, use ViolinC2, ViolinFs4 (*s* often substitutes for ♯). Remember that in Alesis-speak, Middle C is C3. Some manufacturers designate Middle C as C4, so be certain to use Alesis's octave designation. When it's time to double-click the samples to put them into your new Instrument, be careful to add them in precise order from lowest root key to highest root key. (Don't forget that A3 and B3 come after G3, not before C3.)

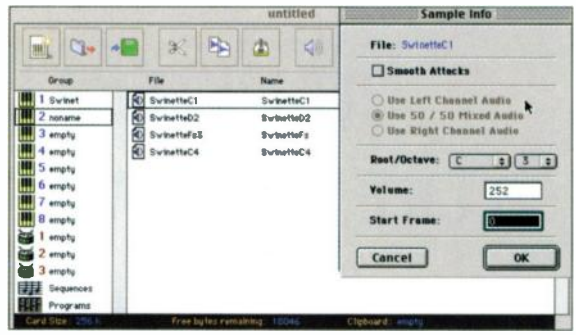


FIG. 3: Once I have loaded my samples into a group, I can set the actual root note and octave of each sample from the Sample Info dialog box.

When you've added all the samples for that Instrument, press the Done button. You'll be asked to name your new Instrument. You will then find yourself back on the main Project page. Double-click on the new instrument and you'll find yourself once again in the Voice Info window. This time the Lower Key and Upper Key parameters will be available to you. You can now create a keymap without having to use a SampleCell editor. Be forewarned: this area gets a little quirky. (I did warn you about in-house software, didn't I?)

Changing Keys

If you single-click on any of the samples you'll be able to set their Lower Key and Upper Key values. However, no sample key range can overlap another key range in that Instrument and each key range must be contiguous. You have to be very clever about the order in which you work or you'll quickly find yourself hopelessly boxed in. This was where I lost the majority of my hair. To add to the confusion, the Lower Key and Upper Key range each have a separate field for the Note name and the Octave number. If you move the Note name first, you'll find that you've violated the "no overlap" rule, even if you were about to enter the correct Octave number. Always assign the Upper Key and Lower Key parameters in the precise order that I'm about to detail.

First, click on the sample with the lowest root key. It should be the top sample in the list. You did add the samples to your new Instrument from lowest to highest root key as I suggested several paragraphs ago, yes? You did include their root keys in their title, yes? Just checking. Set the Lower Key parameters to C and –2. Now go to the Upper Key parameters and assign the Upper Key note to exactly one half-step below the next lowest sample's root key. That one was easy—there were no other samples to get in the way.

Click on the next lowest sample. This next move is important: Change the Upper Key's Octave number from 8 to 7. Now change the Upper Key's Note name to the letter that's one half-step

CHECKING YOUR ALESIS SOFTWARE VERSION

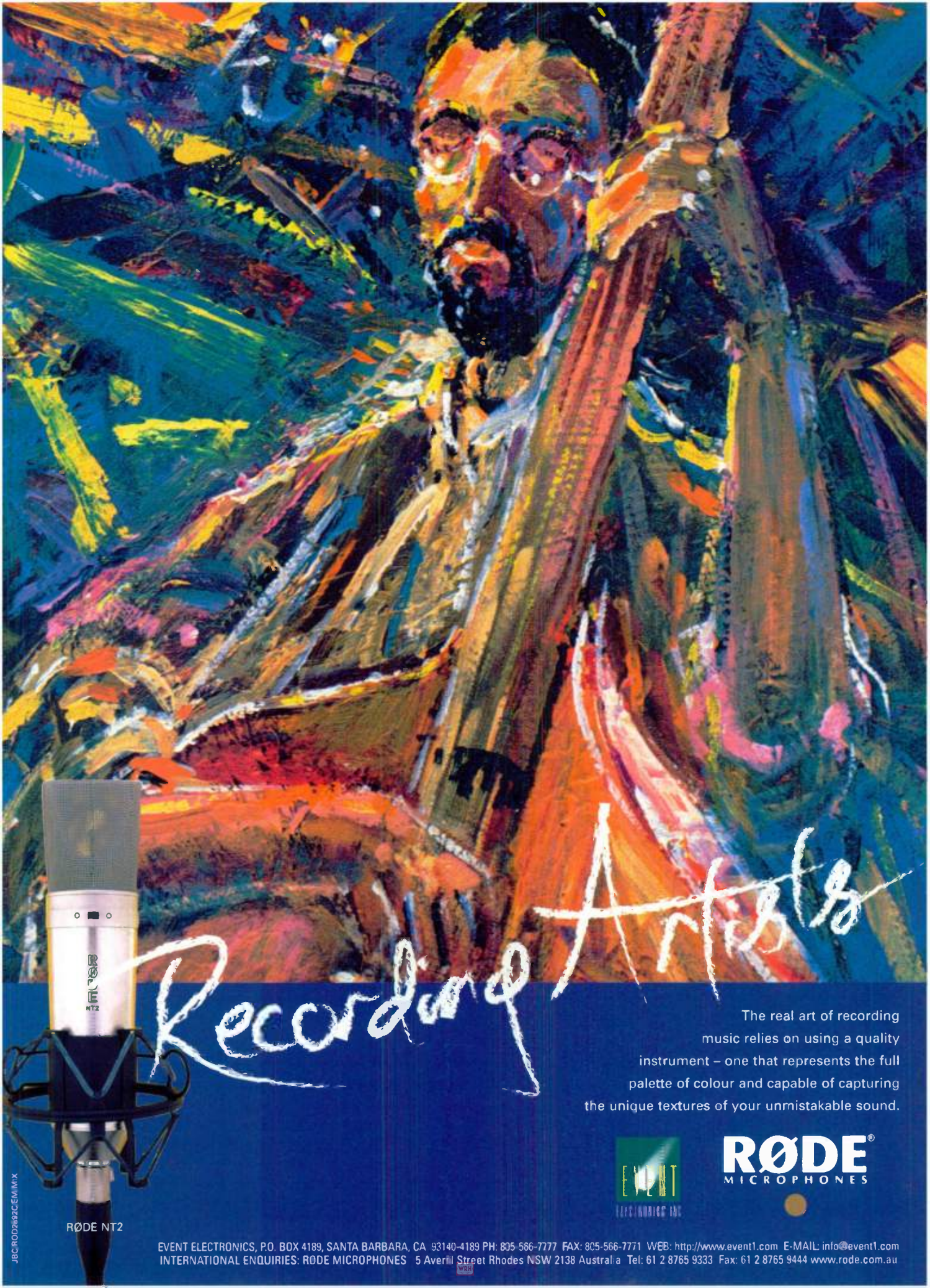
To find out the software version currently running in your QS-series synthesizer or DMPro, follow the instructions below. The software version and date will appear in the unit's display.

QuadraSynth, S4, QuadraSynth+Piano, S4 Plus: Hold down Global and press Program.

QS6 1.00: Hold down Program and press 00 Piano.

QS6 2.00 and higher, QS6.1, QS7, QS8: Hold down Program and press 0.

DMPro: Hold down both cursor buttons.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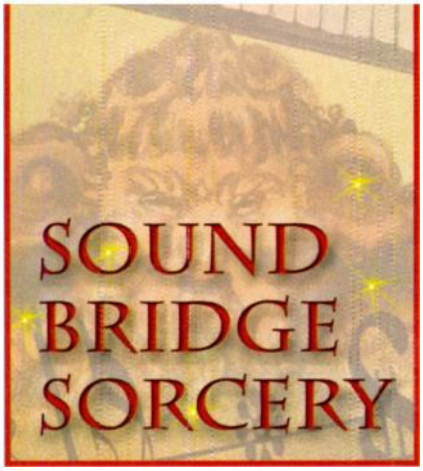

below the third-lowest sample's root key. Only now can you safely set Upper Key's Octave number so that it's just below the next sample's root key.

Now click on the sample with the third-lowest root key. Again, go to the Upper Key's Octave number and change it from 8 to 7. Now change the Upper Key's Note name to the letter that's one half-step below the fourth-lowest sample's root key. Then finally set the Upper Key's Octave number to match the sample's own root key. Continue this exact procedure until you've reached your last and highest sample. On this last one simply let the Upper Key remain as G8.

There is one last step. You still have to double-click each sample so that you can indicate its actual Root note and Octave (see **Fig. 3**). Press the OK button when you are finished. Clicking the Send to synthesizer icon will send your new instrument on its way to the sound card. You can now use this instrument just as you would any other ROM-based instrument in your QS synth.

Sequential Circus

You can add up to 50 Standard MIDI Type 0 and 1 files per sound card. Click the Add Files icon on and use the file pull-down menu to navigate to your source of Standard MIDI Files. Simply double-click on each SMF in the order that you want. When sequences are selected, you can decide if you want the sequence data to be played on the QS or transmitted to external MIDI devices. With your sequence selected in the File window, open the Project menu and select Sequence Info. If you leave the button for Sequence Data via Out Port unchecked, the MIDI sequences will play internally (see **Fig. 4**). When you're through, press Done. The next time you press the Send to synthesizer icon you'll have all your samples, instruments, and Standard MIDI Files ready to go.

To select a sequence on the QS6.1, QS7.1, or QS8.1, just press the SEQ Select button and use the 00 through 40 buttons with a 0-9 button to select any of 50 sequences. If you have two sound cards with SMFs you can use the 50-120 button and a 0-9 button to select those sequences.

FIG. 4: The Sequence Info window allows you to choose if the MIDI data will be played on the QS or external MIDI devices.

For Alesis keyboards other than QS6.1, QS7.1, or QS8.1, see your owner's manual for sequence selection.

Saving Programs

You can save Programs to the sound card via a SysEx file. First, create a bank on your synth that contains the Programs and Mixes you want. You can use internal ROM instruments or the samples and instruments on the sound cards, or any combination that you choose. When your bank is completed, send a MIDI SysEx dump of that bank to the *FreeLoader* program and save as a binary SysEx file.

Using the Add Files icon you can once again go to the leftmost file pull-down menu to get to your SysEx file. Make sure that you select SysEx Files in the lower left Show field of the pop-up window. Double-click on your SysEx file and press Done. Once again you're ready to send everything to your synth with the Send to Synthesizer icon. Remember that each time you send, you're rewriting everything from scratch on the card, so you don't have to erase the card. You will now find that you have an extra bank available when searching with the Bank buttons (to the right of the QS screen).

Playing My Final Card

I hope I've piqued your interest enough to experiment with creating your own sound cards. I promise that it will add tremendous possibilities to your already powerful QS synthesizer and DMPro.

Daniel Fisher *is the product evangelist for Parker Guitars as well as a programmer for Alesis, E-mu, Korg, Kurzweil, TC Electronic, and Yamaha. He blames his success on Sweetwater Sound and his Berklee College education.*

Sound Bridge Compatibility

The table shows all current *Sound Bridge*–compatible Alesis instruments. The number codes in the rightmost column indicate the following capabilities:

1 Indicates ability to work with *Sound Bridge* to compile data (samples, programs, sequences, and so on) to a PC Card sound card via MIDI.
2 The device can utilize samples stored on a PC Card sound card.
3 The instrument can recognize and play program banks stored on a PC Card sound card.
4 Sequences on a PC Card sound card will be recognized and played.

DEVICE	SOFTWARE VERSION	DATA
QuadraSynth	1.0 to 1.07	2, 3
S4 module	1.0 to 1.07	2, 3
QuadraSynth+Piano	2.0 and up	1, 2, 3
S4 Plus	2.0 and up	1, 2, 3
QS6	1.0	1, 2, 3
QS6	2.0 and up	1, 2, 3, 4
QS6.1	1.0 and up	1, 2, 3, 4
QS7	1.0 and up	1, 2, 3, 4
QS8	1.0 and up	1, 2, 3, 4
DMPro	1.0 and up	1, 2, 3, 4

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TRACKING THE MIDNITE

Vultures

Home-recording hijinks with Tony Hoffer and Mickey Petralia, the engineers for Beck's latest album.

Beck's career inspires hope—and a little envy—in most home recordists. His 1993 grassroots hit "Loser" led to the smash 1994 album *Mellow Gold* (DGC), which maintained its popularity throughout the following year. "Loser" is a prime example of how you can make popular music with basic tracks recorded on a 4-track cassette machine.

The song's success gave Beck the freedom to work in any way that he sees fit, and he has explored many musical avenues since first becoming a household name. Directly after *Mellow Gold*, he recorded a lo-fi blues record with indie poster child Calvin Johnson. The album, *One Foot in the Grave*, was released in 1994 on Johnson's love-rock imprint, K Records.

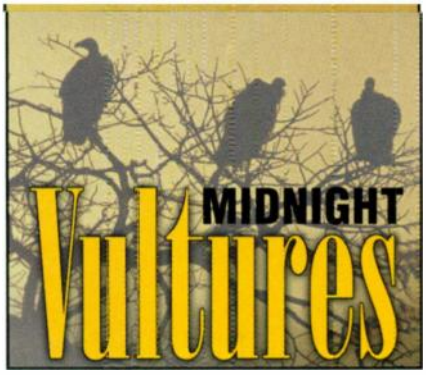
Beck then delved into major studio recording with well-known producers. He released two critically acclaimed albums for DGC, 1996's *Odelay* and 1998's *Mutations*. *Odelay* was a Dust Brothers–fueled monster; *Mutations* was produced with a lighter hand by Nigel Godrich, who is probably best known for Radiohead's remarkable *OK Computer*.

For his latest album, *Midnite Vultures*, Beck wanted to bring the recording process back home. The results are some of his funkiest, most suggestive, and most tongue-in-cheek jams to date. He began by enlisting some long-time friends and touring mates to track songs at his home in Pasadena, California. Several months into the project, he moved to Los Angeles's happening Silverlake suburb, bringing with him not only his belongings, but the recording equipment and tracks as well. So engineers Tony Hoffer and Mickey Petralia, who also produced a few songs on the album, moved their work to Beck's new house. I spoke with Hoffer and Petralia to learn how they captured Vultures' sleazy funk on hard disk.

By Rick Weldon

PHOTOS BY JILLY WELSHELL

Tony Hoffer (left) and Mickey Petralia in their makeshift control room in Beck's Silverlake home. They used the analog Mackie board behind Petralia mainly for monitoring, employing outboard preamps for most of the input signals.



Averill–modified APIs or his Neve 1073s. Unless you're doing drums or a string section, you just need a good pair of mic preamps. Of course, EQs and compressors help.

Mario Caldato, who records the Beastie Boys' albums, always says the same thing: "All you need are a couple of good preamps, one good tube compressor, two EQs, and a good mic, and you're set." If I ever ask him, "Should I use this converter when I'm going from analog to digital?" or "Do I need this compressor?" he comes back with, "The Beatles didn't have that, and listen to their records." It's about using what you have and, more important, it's about the songs.

How did you handle ensembles, drum kits, and other tracks that required a multiple-mic setup?

Petralia: At the Pasadena house we did some horns and drums. Initially we intended to just lay them in as scratch parts, and we went through all of the songs, more or less, in one night.

[Drummer] Joey Waronker came down, and we pieced together a little drum kit. We didn't have a stand for the floor tom, so Justin put on earplugs and held it for a couple of tracks. We got an amazing sound, but that's mostly because Joey is an amazing player. And Joey was locked to the click, so we could loop drum parts by simply lining up an editing point with the click track. That helps so much when you're editing.

Hoffer: Joey takes care to tune the drums well, too, which makes the set record well.

Petralia: Because the drums and horns were in a separate room, we had a lot of loose XLR cables and a snake or two running through doorways. That's the beauty of home recording: you close the door as tightly as you can and go with it. When we compared the original drums from the house with the ones done at a big studio, we often found that the house drums felt and sounded better.

Hoffer: We'd round out the sound in the mix, running individual tracks out to 2-inch tape and adding some subtle EQ and compression. We used [Empirical Labs] Distressors a lot, in addition to Urei 1176s and Teletronix LA-2As.

Petralia: The Distressors are good because they can either mess up your overheads nicely or just kiss something. When we book time in a studio, we usually bring a pair of Distressors, and they always come through in the end.

Hoffer: A Distressor is like a Swiss army knife. It's great for shaping and changing the envelope of any sound.

Petralia: While tracking at the house, we occasionally brought in our own gear, equipment that we had recently bought or had forgotten we owned. I also learned how helpful equipment-rental companies are. Who has $6,000 to spend on a mic?

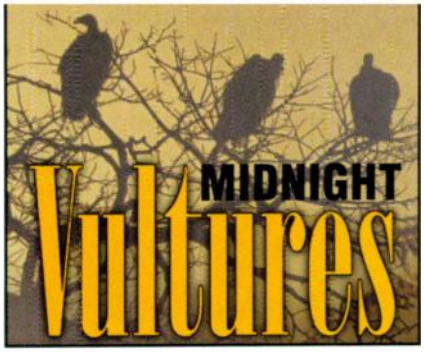
Hoffer: Especially when you're going to use an AKG C 12 only once or twice a year. The rental companies are handy, and more sprout up all the time. For the most part, the people running these businesses are cool and want to help you.

A bank of Hoffer and Petralia's outboard gear lines the wall to the right of Petralia. The rack-mount equipment includes a pair of Brent Averill–modified Neve 1073 preamps and the engineers' beloved Empirical Labs Distressors.

How did you record the vocals?

Hoffer: We recorded many of the vocal tracks with Shure SM58 mics through the API preamps. We did it in the rooms where we had the computers, and the monitors were blaring.

We especially liked the way Beck's vocal sounded through the BLUE



bottle mic, but that mic shows up only on the song "Pressure Zone." We recorded a bunch of songs with it that probably will be released later.

Petralia: The dynamic at the house was constantly changing. Everything was always up in the air and the ideas were flowing wildly, so we had limited time to set up. The SM58s made it easy to get rolling.

Beck and the monitors were in the same room. Did that come through in the vocal track?

Hoffer: Yeah, but the SM58s are superdirectional, so it was never bad. Beck likes to feel the music when he's tracking vocals, which is harder for him when he's wearing headphones.

He would sit on the couch in the den and just start going for it, doing vocals as roughs—scratch vocals. But we ended up using about 90 percent of those vocals on the record. With Beck it's all about the first few takes, and sometimes he keeps the mistakes, too.

So instead of using a mic with a pop filter, he sang straight through a handheld?

Petralia: That's the nice thing about the SM58: it has a built-in pop filter.

Is Beck's vocal the only track on the record with any bleed-through?

Petralia: There was headphone bleed when we recorded acoustic instruments. And Joey hummed the different songs while he played drums. But yeah, bleed was minimal.

You did vocal takes only for a line here and there. Did you match levels to get them sounding the same?

Petralia: We just ballparked it. We didn't punch individual lines, just the sections. For instance, on "Pressure Zone," Beck wanted to give the verse a different vocal tone and character, so we used different mics from section to section.

In music such as this, with things changing drastically every four bars, that probably doesn't matter as much.

Hoffer: It doesn't matter, and people may not even notice. We often make song sections sound completely different from one another, almost like different songs. Listeners won't focus on the vocals' sound quality, but rather on the attitude and vibe of all the elements together. We worked intensively on sections of music and combined them with other sections to create songs.

Petralia: A friend of mine recently listened to the record on his super stereo system. He told me, "I have a new appreciation for what you guys do. I could hear so many things in there." He thought we just turned on the tape machine and started banging on instruments. Once you hear the tweaks and fine touches, you realize what went into it.

It seems as if you double-tracked the lead vocals as a matter of course. How about the background vocals?

Petralia: A lot of people ask, "Who's the woman on 'Get Real Paid'?" That's Beck! *[Laughs.]* He'd do his vocal and double it. Then he'd add a falsetto and a harmony. For songs like "Peaches and

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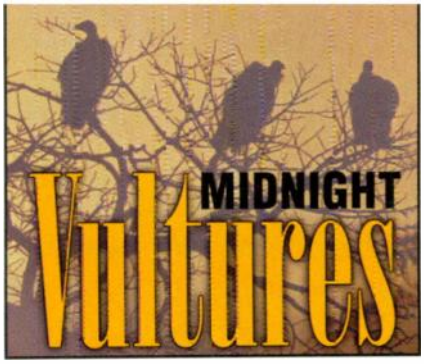


Cream," in which the chorus is sung in unison, everyone—Justin, Roger, Tony, and I—stood around one mic. We did multiple passes and stacked the tracks.

What sequencers did you use?

Hoffer: We started in [Opcode's] *Studio Vision*, but we ran up against a brick wall when we'd recorded too many tracks. We had to wait about ten seconds after we hit Play or Record.

Petralia: We had so many tracks that needed so much processing—Pro Tools was the answer. We sometimes ran 90 or 100 tracks at a time. We probably couldn't have done a project like this using Pro Tools before Digidesign refined its MIDI capabilities, and that's one area in which *Vision* excels.

Hoffer: I still run *Studio Vision* in the background because I use MIDI a lot, and Pro Tools doesn't have an event list editor. I either slave *Studio Vision* to Pro Tools or vice versa, depending on what I want to do.

Petralia: We discussed checking out other programs, but we were so entrenched in what we already had.

Hoffer: We considered porting the tracks to [Emagic's] *Logic Audio* in the middle of the project, but the thought of transferring all the *Studio Vision* stuff was overwhelming. It would have meant learning a whole new sequencer. Beck likes to work superfast, and you need to be three steps ahead of him if the project is going to move efficiently, so the switch was pretty much out of the question.

Were you recording at 24-bit rates?

Petralia: I recorded some stuff at 24 inadvertently. *[Laughs.]* But 24-bit files cause complications when you want to work in [Digidesign's] SampleCell, so we tried to stick with 16-bit files.

Did you use plug-ins a lot?

Hoffer: Not during tracking, but for the Silverlake editing phase, yes. We would use a plug-in to tweak a little section, such as a one-second piece of sound. Steinberg's *Magneto* is cool. So is TC Works' *Chorus*, especially for comb filtering drum sounds. *Renaissance*

Radio control came in handy for Hoffer and Petralia as they mixed in the Silverlake studio.

ompressor, Renaissance Q, Q10—all of Waves' products, really—are good, too. At the time we were beta-testing the new Bomb Factory plug-ins, and those are all great. I'm into the ones that truly tweak stuff, the ones that don't do just one thing.

Right now I'm using BitHeadz's *Unity DS-1*, which I'll take on the road for Beck's tour. I'm going to bring the whole Pro Tools rig, and I'm trying to leave the sampler behind. We can't use SampleCell because we just got a new G4 and we've filled its three slots with the SCSI and TDM Farm cards. It doesn't have room for a SampleCell card, and I don't want to carry a 3U sampler. So *Unity* is great. The program also reads SampleCell instruments, so we can bring a lot of our old songs into *Unity*.

Hoffer and Petralia in the studio. Hoffer is shown sketching out ideas on his cool-as-heck Ibanez IMG2010 MIDI guitar controller.

Did you use any software synths on this record?

Hoffer: We used Propellerheads' *ReBirth* to filter individual sounds. *ReBirth* has a great filter. I'd record a pass of something as I was turning the filter, distortion, or delay knobs, then I'd chop that recording into individual sounds. There's nothing quite like *ReBirth*'s filter and distortion sections. We didn't get a lot of the other host-based applications, such as [BitHeadz's] *Retro AS-1*, until later on in the recording process.

Petralia: And we had Roger Manning.

Hoffer: Yeah, Roger has every analog synth imaginable: Moogs, Oberheims, ARPs—vocoders, patch stuff, and keyboard stuff. Roger has all of the old Roland stuff, too: the TB-303, TR-808, and TR-909. [For more on Roger Manning and his band, the Moog Cookbook, see "In the Kitsch-en with the Moog Cookbook" in the March 1998 issue of **EM**.]

Petralia: He's a living *Retro*. We could hear Roger carrying his eight cases up three flights of stairs.

Did you ever feel the need to leave the house and get some sun?

Hoffer: Yeah. We had a little portable 8-channel mixer that we'd run into the ⅛-inch audio input of a PowerBook, and we'd just use Sound Manager. So we could go anywhere. When we had to get the car washed, we recorded in the car. We brought an SM58 and a book of lyrics, and off we went. We drove through the car wash over and over again and paid the attendant 50 bucks or so. It was fun, and the vocals sounded good. We ended up using them on a couple of songs.

Petralia: They loved us there. They even threw in free Armor All on the tires.

Do you have any other helpful advice for someone working in a personal studio?

Petralia: Get a big hard drive. The price goes down every month. It's best if you're not constantly worrying about how much space you have while you're tracking. A big drive is also nice for backing up. We used to burn CDs for every song; now I just drag back and forth between 18 GB drives to back up.

Hoffer: Don't be afraid to experiment. People think that everything has to be pristine and precise. Understanding the principles of recording is good, but there's nothing wrong with placing your guitar amp face down on the ground and recording its back side. On *Midnite Vultures*, we didn't follow the normal recording practices. Beck works so fast that the best thing to do is to get a track down, without a ton of takes. It was like flying by the seat of your pants—recording an electric guitar with the amp in the control room, no separation, no soundproofing, sometimes even no pants.

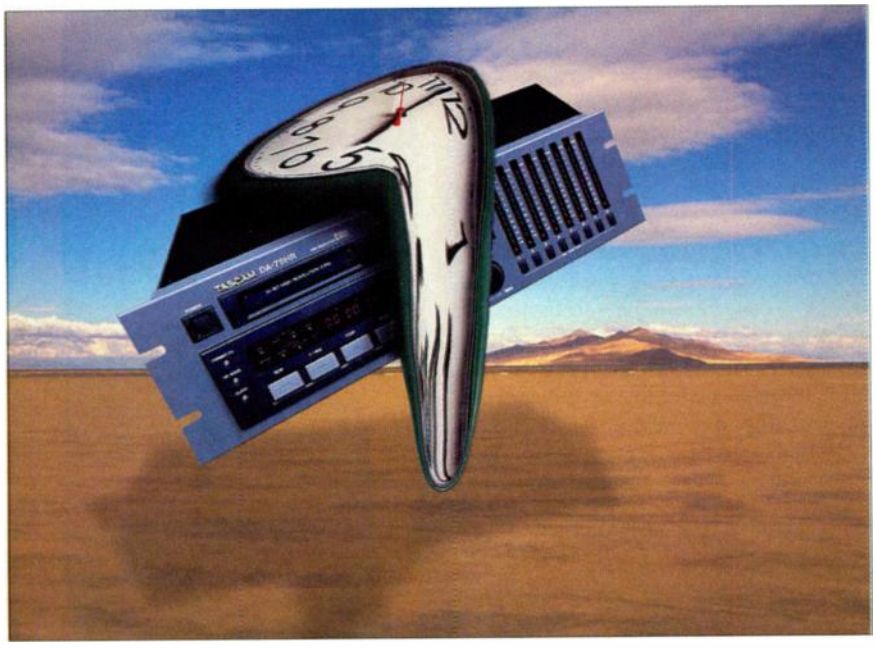
Former **EM** *assistant editor* **Rick Weldon** *has found a name for his home studio that accurately reflects its aesthetics as well as its Bay Area location: Oakland Scavenger.*

Synchronicity

Some thoughts on timing and sync issues from a musical perspective.

By Brian Smithers

In music, as in life, timing is everything, and the technology of synchronization has evolved to address the persistent timing issues that arise from our increasingly complex desktop music systems. If you have more than one piece of gear in your studio, you probably need to start thinking about synchronization. As your studio becomes more complex, you must get a better understanding of how timing information is shared among devices and what level of timing accuracy is appropriate for your needs.



RON BROWN

As you'll see, accurate timing is a context-sensitive notion that begins with the first rule of synchronization: No two clocks are identical, no matter how expensive they may be. World-class studios have to synchronize all of their equipment to a single clock to prevent things from drifting apart—and, to one degree or another, so do you.

YES, MASTER

The second rule is that accurate synchronization depends on all devices getting their timing information from a single *master clock*. Of course, this means that all of your other gear must be able to *slave* to an external clock, usually through a switch or a software checkbox labeled something like "Internal/External Clock." Ideally, you would use the most accurate clock in your studio as the master clock; if you have another device that functions only as a timing master, your options are more limited.

Timing information is carried from device to device by a signal known as *time code*. There are several different types of time code (see the sidebar "Syncspeak"), all of which are commonly lumped together under the term *SMPTE time code*, or just *SMPTE*. SMPTE divides the time line into hours, minutes, seconds, and frames in the format HH:MM:SS:FF. (The term *frames* refers to frames of film and reveals the origin of the standard.) A second is typically

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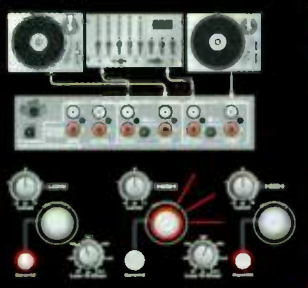
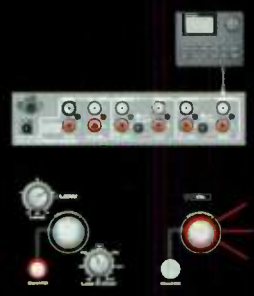
Vocal Remix

Run both decks into EQKiller. Select input 1 or 2, whichever has the vocal lick you wish to isolate. Kill the high and low bands. Use the x-overs to dial in the exact vocal frequency range. Now introduce this isolated vocal into the mix and transform it with EQKiller's mid band momentary button.

Rhythm Remix

Plug a drum machine into input one. Kill the low band and use the X-over to tune out everything but the kick drum. Boost the low band level to +2. Now play the low band momentary whenever you want to punch the kick drum in. Take a basic four on the floor backbeat and jam a totally new groove into it.





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divided into a number of frames ranging from 24 to 30, depending on the film or video format being used.

Simply put, proper synchronization depends on conveying both location and rate. Without location information, two multitrack recorders can't start playback from the same point; without rate information, they will gradually drift apart from each other.

With analog tape machines, synchronization is accomplished by adjusting the speed of the drive motors to counteract any drift from the SMPTE time line. With digital devices, this coordination of speeds is achieved with *word clock*. A word-clock signal cues every digital audio device in the system to record, play back, or transfer each sample at the same time. Variations in the timing of the master device are precisely duplicated in every slave device.

CLOSE ENOUGH FOR JAZZ?

Horticulturists will tell you that a weed is just the right plant in the wrong place. The same could be said of a wrong note. The question is, how far out of place does a note have to be before it becomes "wrong"? If members of an orchestral violin section sneak into a quiet note over the span of a fourth of a second, the note will blossom beautifully. However, that same quarter-second discrepancy between a trumpet and a tenor sax can turn an intricate bebop line into a chaotic echofest.

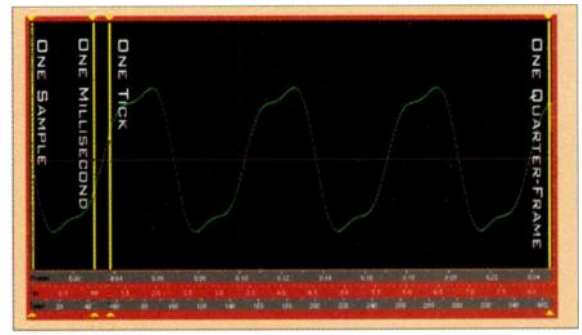

FIG. 1: It's important to understand the relative sizes of the time units used in synchronization. This diagram assumes a sample rate of 44.1 kHz, a frame rate of 30 fps, and MIDI resolution of 480 ppqn at a tempo of 100 bpm.

Synchronization technology enables us to guarantee accuracy in terms of picoseconds (millionths of a millisecond), but when does that kind of precision matter? It certainly doesn't in a typical multitrack session. Part of what makes live musicians sound alive is the subtle interaction of their minor imperfections. That's one reason why quantized MIDI sequences often sound overly mechanical: the parts line up too precisely. To counteract this excess precision, most sequencing software now features algorithms for "humanizing" (randomizing) a sequence and offers degrees of quantization for fixing only the most egregious rhythmic errors.

The more definite a sound's attack, the more it suffers from timing errors. If you record two drummers playing the same part, timing discrepancies introduced during the recording or playback process will be more obvious than with vocals or strings. To see for yourself just how forgiving our ears are of timing "errors" within an ensemble, try this experiment. Open your sequencer

and record a simple drum or percussion part, such as a scale exercise on a marimba patch. Now copy it into another track and assign the new track to a xylophone patch. (Be sure to assign this track to a different MIDI channel.) Both parts should play back in perfect unison, as if the two timbres were layered at the patch level.

Now slide one part a tick or two later and see what happens. The results, of course, will depend on the tempo and the MIDI resolution you're using; but after a couple of ticks, you'll start to hear some "flamming" of each attack. As you slide the parts further away from each other, you'll hear each one more and more distinctly from the other, until at some point they cross the line and end up just sounding sloppy. Now change both parts to string patches and see how much further apart you can slide them before they sound wrong. Your ears will probably forgive about twice the discrepancy between string parts as between percussion parts.

If you're trying to sync a digital audio sequencer to your MDM for typical multitrack sequencing and recording, then perhaps you don't require the highest-resolution synchronization hardware on the market. In fact, countless major recordings from the past decade have relied on MIDI Time Code (MTC) for this sort of arrangement. MTC is accurate to a quarter of a frame, which translates to a maximum error of about 8 milliseconds (see **Fig. 1**). That may sound like a lot, but consider two things: first, the typical margin of error is significantly less; and second, the errors in subsequent overdubs aren't cumulative.

Why does synchronization matter at

SYNCSPEAK

black burst Synonymous with house sync, its name derives from the fact that it is a video signal with no picture, which would yield a black screen if displayed. Also known as *video sync*.

frame rate The number of frames of film or video displayed per second. Film runs at 24 fps, video at 29.97 fps. Most music-only production is done at 30 fps.

house sync A video signal used as a master timing reference for video and audio devices. Like word clock, it conveys rate information but not location information.

LTC Longitudinal Time Code, the most common form of SMPTE in audio applications. The time code is converted into a modulated audio tone similar to modem noise and then striped (recorded) to one track of an audiotape recorder. The playback of that tone is subsequently read by a synchronizer, which controls the speed of slave devices.

MTC MIDI Time Code, a form of SMPTE that can be transmitted over MIDI connections.

SMPTE time code The now-ubiquitous format of hours:minutes:seconds:frames adopted by the Society of Motion Picture and Television Engineers for conveying location and timing information to video and audio devices. Also known simply as *SMPTE*.

Superclock Digidesign's version of word clock that runs at 256 times the sample rate for extra precision.

VITC Vertical Interval Time Code, a form of SMPTE commonly used in video applications. A video frame is drawn in two interlaced passes of the cathode ray gun. The point at which the gun resets itself from the bottom corner to start over at the top corner is called the *vertical blanking interval*, and time-code information is inserted at this point. The time-code display window on a video screen is derived from VITC.

word clock The signal that defines the precise timing by which each sample is recorded, played, or transferred. Unlike time code, word clock doesn't carry location information.

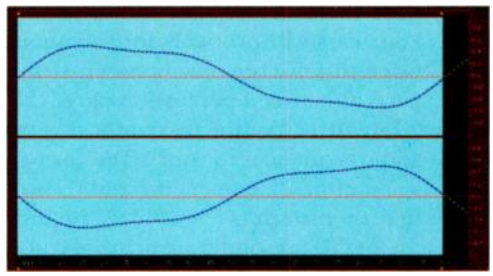

FIG. 2: One complete cycle of a 440 Hz waveform takes roughly 100 samples at a sample rate of 44.1 kHz. If you move one side of a stereo pair of such waveforms about 50 samples in either direction, the two sides will become 180 degrees out of phase and effectively cancel each other out.

all? After all, once an audio file is in your computer, you're going to drag it where you want it anyway. In fact, if you mix all of the audio in your computer, sync doesn't matter. But if you want to lay edited audio back to your MDM, you'll want to return it to where it came from with reasonable precision, and MTC does this adequately under many circumstances. (It pays to be resourceful, though. Once, caught without a viable sync arrangement, I re-recorded an edit back to tape—on the fly—and then nudged it into place with the recorder's track offset.)

CLOSE DOESN'T COUNT

Lest you think that synchronization is all hype, consider the task of fixing a digital glitch in the left channel of a stereo mix. You dump the single track from tape into your audio-editing program, clean up the problem using the Pencil tool, and lay the track back to tape. The result is chorusing, flanging, phase cancellation—you name it. Stereo pairs are extremely unforgiving of timing errors between channels.

If you open any stereo mix in your editor and drag one channel more than a few samples in either direction, the discrepancy is immediately apparent. Do a little math, and it's easy to see why. One complete cycle of a waveform at 440 Hz (the standard tuning A) takes about 100 samples, or 2.27 milliseconds. Drag the wave 50 samples in either direction and you've achieved complete phase cancellation (see **Fig. 2**). It takes only a fraction-of-a-millisecond discrepancy to create pronounced chorusing.

If you need to work within such tight tolerances, you'll want the ability to sync your gear to within at least a couple of samples. With the right combination of gear, you can even get sample-accurate synchronization. This is the purpose of ADAT's proprietary 9-pin sync connection: it enables you to achieve single-sample positioning accuracy. Whereas older DA-88s need a separate connection for sample-accurate sync, the TDIF connection carries all the required data.

Even lacking sample-accurate sync, you can move tracks back and forth without creating phase problems if you're resourceful. Using the aforementioned example, you could transfer both channels at once (instead of just the problematic left channel) to the computer for editing. They would arrive still in phase, and if you didn't change their relative position they would stay in phase when you transferred them together back to tape. They might end up offset by a few samples from their original position on tape, but if you choose your edit points according to phrase structure, this discrepancy shouldn't be noticeable. If your audio interface has enough inputs and outputs, you can even transfer an entire multimicrophone set of drum tracks at once.

LIKE CLOCKWORK

Synchronization doesn't have to be a nightmare, and it doesn't necessarily require expensive gear. When you understand the timing issues of different musical contexts, keeping everything in sync is not that difficult. Ultimately, it boils down to a few points:

1. Use your most accurate clock source as the master timing reference.
2. Slave all other devices to the master clock.
3. Use sample-accurate sync to prevent phase problems between similar audio content.
4. Don't be afraid to rely on MTC in most situations.
5. Be resourceful; cheat whenever necessary.

Above all, use your ears. If it sounds right, it is right. Remember, it's all in the timing.

Brian Smithers *is searching for a master clock to sync his work/sleep cycle with Earth's day/night cycle. Contact him through his Web site, members.aol.com/notebooks1.*

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RECORDING MUSICIAN

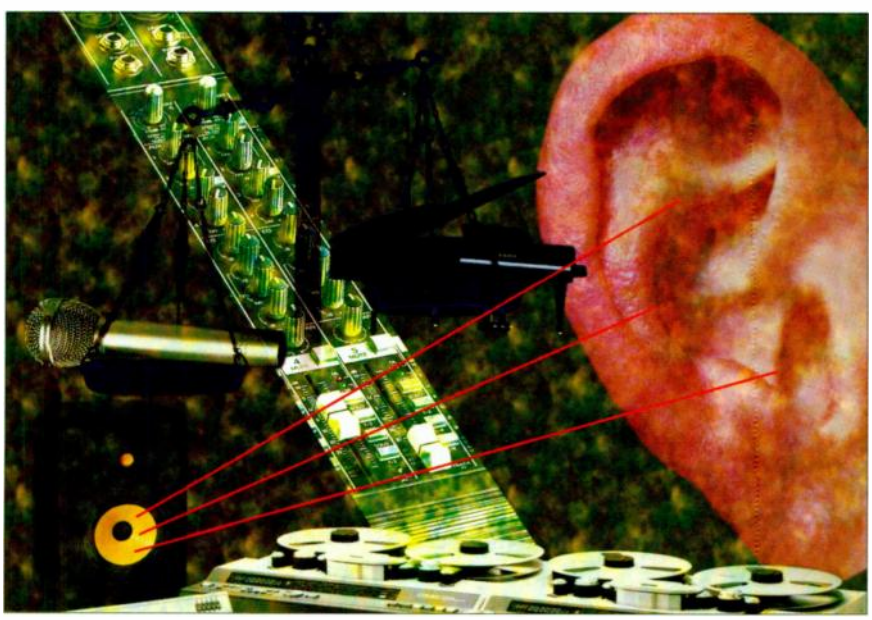
Ten Tips for Nailing a Mix

There's more to mixing than meets the gear.

By Brian Knave

Mixing, like any art, can't be reduced to a formula. All that counts is the finished product, which means there is, literally, any number of ways to proceed. Just the same, I have developed a somewhat systematic approach to mixing that usually works for me and for the types of music I tend to mix (the standard rock instrumentation of drums, bass, acoustic and electric guitars, keyboards, vocals, and so on).

In this column I'll share some of the thoughts and techniques that go into my approach, including a few gleaned from friends and colleagues. None is radical, nor is any writ in stone as *the only way;* indeed, experienced mixers will probably be familiar with most, if not all, of my methods. But if you're relatively new to mixing, or if you consistently struggle with particular problems in your mixes, you may find something of use here.



STEPHEN RAMIREZ

IT'S A STYLE THING

Mix appropriately for the musical style. That may sound obvious, but what we're really talking about is musical *vision*—that is, having an idea, in advance, of how the final product should sound. Essentially, mixing is about getting the right balance of levels. Just as different styles of music employ different instrumentation, the relative volumes of the different instruments vary from style to style. Therefore, it's important to be familiar with the style of music that you're mixing, to know which instruments get foregrounded, which backgrounded, and so on. Of course, it's okay to break the "rules"—but it sure helps to know them first.

For music with vocals, the level of the lead vocal is typically a defining element of the mix style. In heavy rock, for example, vocals tend to get mixed low, as if they were just one of the instruments—indeed, they may even be mixed lower than the main guitar tracks (check out Rage Against The Machine). Commercial country music,

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on the other hand, tends to feature the vocals; no instrument is allowed to distract too much from the voice and lyrics. Vocals are prominently featured in pop, too, yet slightly less so, on average, than in country.

Other defining elements of certain mix styles are drum and bass levels. In jungle, house, and other "electronic" dance styles, for example, an emphasis on the bass and rhythm elements is common. Consider, too, how drums and bass are featured in styles such as reggae (especially the modern stuff), hip-hop, and, well, drum 'n' bass. Classic rock 'n' roll, on the other hand, often showcases electric guitars or keyboards.

THE BIGGER PICTURE

When you start work on a new mix, rather than solo the individual instruments and commence equalizing, compressing, and so on, try first to make the song sound great using only fader levels and pan positions. This approach—the art of mixing stripped to its essentials—will not only get you focused right away on the bigger picture, but it will also help you hear what's working and what isn't. For example, you may find that certain instruments are fighting for the same sonic space. In that case, try panning them far apart in the stereo field. Later, you can differentiate them further with EQ or what have you; for now, though, keep working to make the song sound as good as you can using only levels and pans.

One secret to a great mix is knowing how to select and highlight the one or two instrumental elements that make the song rock hardest (an aesthetic choice that usually relates, by the way, to the aforementioned style considerations). Once you determine the element (or elements), the rest of the mix can be built around it. In other words, great mixes are not usually "democratic," in the sense of each instrument having an equal say, but instead tend to employ a hierarchy of levels.

In vocal-based songs, the challenge often is to find the right balance between the foregrounded elements and the lead vocal. Instrumental-based songs, on the other hand, are usually a bit easier to manage, as typically there is only one instrument to spotlight.

SQUEAKY CLEAN

Clean up individual tracks with mutes, gates, or however. To do this, you have to solo each track (in particular, those with instruments that come in and out of the mix), listen carefully for any unwanted noise, and then squelch it. If you have automation (either digital or analog), programming mutes or gates is the quickest and easiest approach. If you mix without automation, you can use outboard gates and/or expanders. As a last resort—say, if you don't have enough gates to go around—you can always erase sections between parts. (This is risky, so make a copy of the track first in case you screw up.) The goal, of course, is tracks that are free of extraneous sounds—coughing, lip smacks, chair squeaks, and so forth.

You should also check for unwanted noise on continuous-playing tracks—for example, amp hiss on an electric-guitar part. In the case of amp hiss (or other constant noise), try patching in a single-ended noise-reduction unit, such as Behringer's Multiband Denoiser SNR2000 (see **Fig. 1**).

THE GREAT EQUALIZER

A mistake that novice mixers often make is boosting every band of EQ on every channel. Though this may initially make the signals sound "better"—at least to inexperienced ears—the seeming improvement is largely due to the signal being louder. (Remember, boosting EQ adds gain.) But there are other reasons not to go this route. Not only does it fairly defeat the idea of *equalizing*, but on analog mixers—at

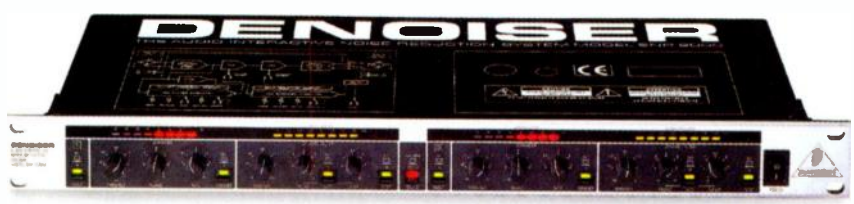

FIG. 1: In addition to two channels of single-ended noise reduction, the Behringer Multiband Denoiser SNR2000 also provides an expander/gate on each channel.

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least those typically found in personal studios—it also usually adds circuit noise to the mix. Indeed, with every band boosted on every channel, the compounded noise can be considerable. (To hear for yourself, stop the tape, turn up the monitor-room output, and listen while switching the channel EQs into and out of the signal path.)

Whenever possible, then, cut rather than boost—or use a combination of cuts and boosts—to get the sound that you want. Cutting lets you eliminate undesirable aspects of the signal without adding noise; afterward, if necessary, you can add make-up gain with the fader. In general, experienced engineers end up using a combination of cuts and boosts on any given channel.

For those unfamiliar with subtractive EQ, it's instructive to spend some time experimenting with sweepable mids. First, dial in a radical cut—say, 15 dB. (If you have control of the bandwidth, set a narrow Q, as well.) Now, turn the sweep knob slowly from hard left to hard right while listening to how the cut affects the signal at different frequencies. Next, do the same thing with a 15 dB boost and compare. What you're listening for is how a cut on one side of the frequency spectrum results in a sound quite similar to that provided by a boost on the other side. For example, a low-mid cut (depending on the source, of course) can result in an apparent brightening of the signal—much like what you would get by boosting the high mids.

Here's another important point: soloing an instrument to EQ it can be helpful, but the only thing that really matters is how the instrument sounds in the mix. Therefore, be prepared to change the EQ once all the other instruments are brought in. Often, after an instrument is equalized appropriately for a mix, it sounds quite unappealing when soloed.

Still another trick is to EQ the effects rather than the instrument itself. I find this to be especially helpful when I'm happy with the quality of the signal (that is, how well the instrument was recorded) and I want to alter the tonal balance only subtly, without messing up the naturalness of the sound. You can EQ the effect either inside the effects unit (advantageous, because digital changes don't add circuit noise) or by patching the return into a channel strip and equalizing from there (which is nice, because it allows you to

For manufacturer information, please see Contact Sheet, p. 205.

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readily pan the signal or even compress it, if you want).

ROOM WITH A BLEND

Choose effects carefully and limit the number of like effects. For example, few mixes need more than three different reverbs. Indeed, numerous hit songs have sounded great using only one.

The first thing I look for—especially if the song was built overdub by overdub in a makeshift personal studio—is a global reverb that complements the whole mix. The idea here is to create the illusion of everything being in the same space. Halls and rooms are good starting points; in general, it's best to save the more unusual effects for individual instruments. Typically, I send every channel to the global reverb (except, perhaps, the bass guitar's). To hear what I'm doing, I initially crank the effects return, and I may even solo it to better hear the size, decay, and overall coloration (dark or bright) of the space. Once I've dialed in the most complementary settings, I turn down the returns to the desired level.

Just as it's helpful to spotlight a particular instrument or two in the mix, it's usually a good idea to apply supplemental effects in a similar fashion. That is, don't put a different, outrageous, and equally glaring effect on every instrument in the mix. Instead, pick one or two elements to focus on and concentrate your more creative use of effects there. You don't want to confuse the mix with too many competing elements.

When using multiple reverbs, another thing to watch out for is phase distortion and cancellations from multiple stereo returns. One of the worst offenses can come from accidentally reversing the left and right return cables from one of the processors going into the board. Talk about a comb-filtered jumble! But even with everything properly cabled, it's not uncommon to get masking and other strange interactions when combining stereo reverbs (whether they are true stereo or not).

Fortunately, it's easy to test for this problem: simply A/B the mix in stereo and mono. Going to mono almost always causes some reduction in stereo effects—often a great deal. But the thing to listen for is the phasey stuff or the disappearance of critical content. If you find a problem, try systematically returning each effect in mono (that is, by disconnecting one of the stereo return cables—starting with the right, as left is typically the preferred mono connection). This process of elimination should turn up the worst offender. Once you've identified which box is the problem child, you'll probably find that you can get by just using it in mono, especially in a dense mix. In fact, some purists make it a point to use only one stereo effect in a mix; the rest are returned in mono.

LEVEL HEAD

After dialing in all the compression, EQ, and effects, return your attention to the overall balance of levels. In the end, this is what will make or break your mix. There's an art to perceiving how best to balance levels. A killer mix has an energy that seems to jump out of the speakers. It feels visceral and sounds loud—no matter what the volume. All of the instruments and elements can be heard, yet some illusion is going on in that the foregrounded elements don't sound foregrounded; they just fit into the mix perfectly, sounding neither too soft nor too loud—sort of like when an image comes perfectly into focus through the lens of a camera. Such is the power of a well-arranged hierarchy.

As you get closer to finalizing settings, note that the slightest fader move can cause a significant change in the balance of elements. For example, say you've brought up the lead and rhythm guitar tracks and found a really great balance between them and the lead vocal. But now, listening back, you think the drums sound a little low—they're just not quite as slamming as before. So you push up the kick and snare a bit, say 1 or 2 dB each. That doesn't sound like much, and yet suddenly the whole mix is thrown off because the guitars are no longer out front.

Mixing, you see, is a very delicate matter. So rather than pull more things up to compensate, figure out what you brought up too much and nudge it back down. (Note, too, that as the majority of

MAGIC BLACK BOX

Though it might seem odd to tout a piece of gear in a column about mix tips, I would be remiss not to mention a unit that graces nearly all of my mixes and truly helps them sound better. That unit is the BBE 462 Sonic Maximizer. (Note that the 462 was recently replaced by the 482, which costs $349 and is said to sound smoother and warmer and have more headroom. The higher-end, balanced version is the 882, at $599, formerly the 862.)

The Sonic Maximizer provides two different processes: linear phase shift and dynamic EQ. I often use a bit of the EQ, but for me it's the phase shifting that makes the difference. The Maximizer is said to restore harmonics to their original time arrangement—a relationship that gets "flattened out," so to speak, by speakers. The Maximizer remedies this flatness by introducing a linear phase shift across the entire frequency spectrum (20 Hz to 20 kHz), effectively slowing down the lower frequencies so that higher harmonics reach the ear before lower ones. The result is a clarification of detail and an improved sense of dimensionality; after engaging the BBE Process, as it's called, I can better hear "around the edges" of the instruments.

BBE Sonic Maximizers are extremely easy to use. But the 462, 862, and previous models employed a somewhat counterintuitive "Lo Contour" control for which unity-gain was not indicated. On those units, unity-gain for Lo Contour was around 11 o'clock—a design that allowed for both low-frequency cutting and boosting. However, the newer models have dispensed with this design and no longer provide the variable low-frequency cut (which no one seemed to use anyway).

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FIG. 2: Coleman Audio's versatile LS3 Balanced Line Level Selector is a high-quality yet inexpensive solution for switching between two or three pairs of monitors. It employs passive switching and features balanced ¼-inch TRS input and output jacks.

faders cross the nominal point, it becomes increasingly important to pull some down as opposed to continuing to push them all up, or else, before you know it, you're the victim of *fader creep*—that insidious state in which all of the faders have crept to the top of the board!)

POINTS OF REFERENCE

One of the most helpful things you can do—especially if you're new to mixing—is compare your mix to songs from a stylistically similar commercial CD. Though this can be done at any point in the mix, I usually wait until late in the game, if only to see how far I can go on my own. On the other hand, if you don't have an immediate sense of direction for a mix, listening to a few finished mixes may give you some ideas.

Be careful when referencing, though: there's really no standard in mixing, so your mix could sound quite different from someone else's and still be very good. For me, the point of referencing is to make sure I'm in the ballpark—not to tell me how to throw the ball. I find referencing to be most useful when making final decisions about levels for bass, vocals, and effects.

QUIET ON THE SET

As we've all heard time and again, it's important to monitor at various levels, including loud, soft, and a couple of points in between. But the single most helpful monitoring trick that I've learned is to listen not only at very low levels (75 dB or less), but also in mono. Hearing the mix quietly from a single point allows you to really home in on levels and the overall balance of elements (not to mention how well you have succeeded in differentiating between those instruments that were initially competing for the same sonic space). When you can hear everything clearly this way, you know you're close to nailing the mix.

The "quiet mono" trick is also a great way to monitor if you're having to ride levels manually (and let's face it, even automated levels are done manually at some point). In fact, I often run my mixes this way while printing to tape. Of course, only the board is in mono—the signals going to the record deck are still stereo.

THE MORE THE MERRIER

The ultimate test of a mix is that it should sound great no matter what it's played back on. Therefore, the more speakers you can monitor with, the better. This can be an expensive proposition, but it doesn't have to cost an arm and a leg.

The cheapest solution I know is to use a boom box as your secondary monitor source. In my studio, I mult the mixer's stereo outputs through a half-normaled patch bay and run one stereo pair to my DAT deck and a second pair to the CD inputs on the boom box. When I want to monitor through the boom box, I simply turn down the control-room volume, flip the boom box to CD mode, and adjust the output from the board accordingly. Clients appreciate this, too; it lets them hear how their songs will sound "in the real world." (The real world *is* a boom box, right?)

If you can spring for an extra pair of speakers and an extra power amp (or a second set of powered monitors), a tidy and convenient solution is to install a line-level selector between the mixer's outputs and the inputs to the power amps (or powered monitors). The Coleman Audio LS3 Balanced Line Level Selector (see **Fig. 2**) costs only $120 and provides three sets of outputs for this type of application. And if you're looking for a pair of small, inexpensive secondary speakers, check out Radio Shack's Pro-X44AV monitors ($59.99 each), formerly known as Minimus 7 and then as Optimus 77. These are great-sounding little units, and I know several pro engineers who swear by them.

Of course, the multiple-monitoring technique needn't stop with the speakers in your studio. Once you've printed

A TRUE WORK-AROUND

Here's a trick I learned from George Petersen, editor of *Mix* magazine. I had spent many weeks recording a bunch of tracks using really fine mics and preamps. To get the best sound, I bypassed the board and recorded direct to tape (ADAT). I was meticulous, and I spent lots of time getting things just right. Everything went to tape sounding great—clean, crisp, and really tight and defined.

Once I had mixed everything together, though, I noticed a loss of clarity. To pinpoint where the loss was taking place, I subtracted things from the mix one by one—effects, EQ, compression, and so on. But none of those maneuvers remedied the situation. I finally began to suspect that the mixer was muddying my signals.

Petersen recommended that I bypass the master-output section of the board. After all, he explained, at best there are only two op amps in the master output bus (and in a budget mixer, none of the op amps is that great to begin with). So I bypassed that section of the board and noticed an improvement right away. Granted, it was a subtle improvement—but any improvement in clarity is better than none at all.

Bypassing the master-output section of your mixer is simple—that is, as long as the mixer provides stereo-master insert points. Simply remove the cables from the ¼-inch stereo outputs (the ones that go to your 2-track record deck) and insert them halfway into the stereo-master insert points. (It's critical that they be inserted only halfway, or else no signal will get through.) Of course, this means that the master fader no longer has any effect. But as long as you're mastering (or editing) the finished mix later, you can do your fades then. (Note that it helps to have a variable input control on your 2-track record deck for this application.)

a final mix, run off a cassette copy or burn a disc, and go listen to it in your car stereo, your home stereo, your Walkman, and anywhere else you can play it. (I've even known people to take their mixes into high-end consumer-audio stores and listen to them there.) And don't forget the "around the corner" trick. That is, listen to your mix from one room while it's playing in the other—a technique that can reveal imbalances you overlooked during hours of close-field monitoring.

Time is an important buffer, as well. When I'm doing a critical mix (that is, of a song that's going to be on a CD), I always try to allow a day to "sit on the mix." I send everyone home with a copy (including myself); we all make notes about what we like and don't like; and the next day we come back, make final tweaks, and print the mix again.

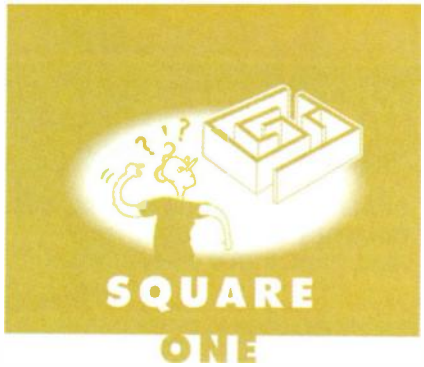
INCREASE YOUR ODDS

Just as photographers will shoot several rolls of film to get one usable shot, you owe it to yourself (or your client) to print several versions of the final mix. That way, you increase your odds of getting the "perfect" mix. Essentially, this is another way to increase the "time buffer," so you can wait until mastering time to choose the mix that works best for the overall project. There's nothing like some perspective to help you make the best decision.

I usually print at least four versions of a mix. The first is the one that I think is right. The second is the same as the first, but with a bit more bass. The third is the same as the first (that is, I pull the bass back down), but with a bit more vocal. And the fourth has both more bass and more vocal. In addition, if there's time, I may print four more mixes using the same approach, but each with more effects. (I tend to go light on effects; however, if you like to mix heavily with effects, you may want to print extra versions with the effects returns pulled down a bit.)

Be sure to keep clear notes (including start and stop times) about what distinguishes each mix. Professional mastering is expensive. You don't want to show up at the mastering house with twelve different mixes of each song, but no clue as to which is which!

Brian Knave *is an associate editor at* **EM.**
Special thanks to Barry Cleveland and George Petersen.



Multi-Effects 102

Continuing education in multi-effects processors.

By John Duesenberry

Last month, I discussed the basic characteristics of multi-effects processors, focusing on delay, phasing, and reverberation effects. In this month's column I'll continue my survey of different types of effects commonly featured in modern effects processors, including distortion, pitch-shifting, and amplitude modulation. I'll deal with dynamics processing another time.

Though the focus of this article is on hardware, many of the processes I'll describe can also be found in software plug-ins or stand-alone applications. Many of these plug-in effects offer real-time tweaking, but regardless of whether they work in real time, their underlying concepts are similar to those of their hardware counterparts.

JACK DESROCHER

ACCENTUATED AMPLITUDE

A number of common effects are based on *amplitude modulation* (AM), a technique in which a *modulator* signal controls the amplitude of an input, or *carrier*, signal. *Tremolo*, for example, is a pulsating effect that has been popular since it was introduced in guitar amps in the 1950s. It's produced by a sine or triangle wave modulating the amplitude of an audio carrier. *Autopanning* sweeps a signal back and forth between stereo channels, under the control of an LFO. Usually, tremolo and autopan effects have similar parameters: rate, depth (amount of effect), and LFO waveform.

In *AM synthesis*, the modulator and carrier signals are both in the audio frequency range (approximately 20 Hz to 20 kHz). AM synthesis generates *sideband frequencies* in the output spectrum. Sideband frequencies are equal to the sums and differences of the input frequencies. Often, the sidebands are not harmonically related to each other, producing distinctive clangorous timbres. There are several forms of AM synthesis. One of the most popular is *ring modulation*. A true ring modulator produces only sidebands, suppressing the original input signals. For an in-depth

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Juan's photo by Eduardo Patiño

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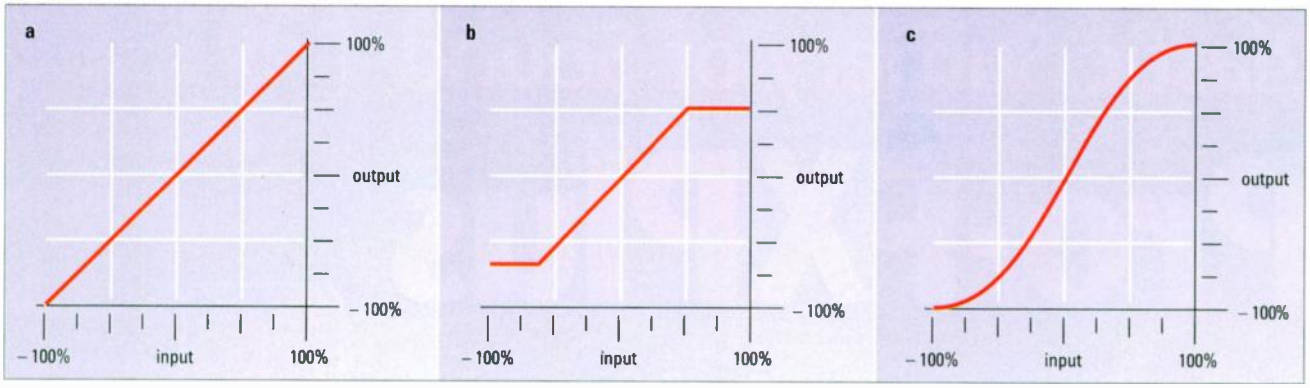

FIG. 1: Transfer functions can be visualized as plots on a two-dimensional graph. Pictured here are the functions that would be used for (a) a linear device; (b) a hard-clipping device; and (c) a soft-clipping device.

discussion of amplitude-modulation techniques, see "Square One: Modulation Synthesis Methods" in the March 1999 issue of **EM**.

TRANSFER, PLEASE

To understand how distortion devices work, it is useful to understand the concept of a *transfer function*. A transfer function maps a device's input amplitude values to its output amplitude values. By plotting inputs against outputs, you can visualize the transfer function of a given device. **Fig. 1a** shows a perfect *linear* transfer function, in which input values map to identical output values. A device with a linear transfer function may amplify, invert, or attenuate a signal (in **Fig. 1a** the signal is passed through untouched), but it doesn't modify the signal's waveshape. Therefore, a linear device doesn't introduce any spectral changes. An ideal amplifier would have a completely linear transfer function.

Fig. 1b shows the transfer function of a device that is linear through part of its input range, but that will not exceed certain positive and negative levels. The effect of this nonlinearity is that the signal is *clipped* when its value exceeds the linear range (see **Fig. 2**, top track). Clipping is a simple example of *nonlinear waveshaping*.

This *hard clipping* introduces odd-numbered harmonics into the spectrum. Hard clipping is characteristic of solid-state devices, which is one reason why many guitarists abhor solid-state amps. (Paradoxically, many tube-loving guitarists buy fuzz boxes, which are nothing but simple hard-clipping devices, and put them in the signal path before their tube amps!)

Hard clipping is by no means the only type of nonlinear waveshaping. **Fig. 1c** shows a nonlinear transfer function that distorts the signal in a more subtle way. The output becomes more distorted as the input approaches maximum positive or negative amplitude. The transition from undistorted to clipped signal is gradual, so the sonic effect is much less jarring. Distortion produced by this type of "S-shaped" transfer function is often called *soft clipping*.

The bottom track in **Fig. 2** shows a soft-clipped sine wave. Soft clipping is characteristic of tube devices and lends "warmth" to the signal. The voltage-controlled amplifiers (VCAs) employed in many Moog analog synthesizers had S-shaped transfer functions, although they were not tube devices. The famous "fatness" of the Moog sound was partially due to these VCAs.

In multi-effects processors, distortion effects usually involve some form of hard or soft clipping. But a distortion processor could use any imaginable curve as a transfer function. The best way to explore this is to use a software-based processor that lets you draw an arbitrary transfer function and apply the curve to your audio input. Digidesign's *Turbosynth SC* and Cycling 74's *Pluggo* package (both for Macintosh) include waveshaping processors, as does Sounds Logical's *WaveWarp* for Windows.

SWITCH PITCHER

For decades, musicians and engineers have sought ways to alter the pitch of a signal without changing its duration or introducing significant distortion or other artifacts. This Holy Grail has yet to be found. However, technology has progressed to the point where you can expect a midrange effects box to achieve fairly clean pitch transposition over a limited range.

Real-time pitch-shifting is usually accomplished using some variant of the following technique. Samples are read into a memory buffer at your system's sample rate and read out at a different rate—faster to raise the pitch, slower to lower it. To preserve the duration of the signal, samples are either repeated (if the pitch is being raised) or dropped (if the pitch is being lowered). This process, needless to say, can introduce nasty discontinuities and other artifacts into the signal; these become worse as the pitch-shift ratio increases. Designers of pitch-shifting devices have devoted considerable ingenuity to reducing pitch-shift artifacts, devising algorithms that adapt to signal behavior.

Most pitch-shift processors offer a range of 1 or 2 octaves, but the clean range will often be quite a bit smaller. You should listen critically before buying: pitch-shift quality varies considerably

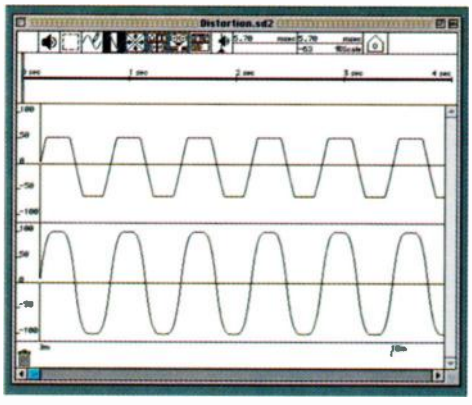

FIG. 2: Clipping will occur when a signal's output exceeds the audio system's dynamic range. The top track shows an example of hard clipping; the bottom shows an example of soft clipping.

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One of the most common pitch-shift applications is *chorusing*. Acousticians long ago observed the "chorus effect" that takes place when a group of similar instruments (such as a violin section) plays in unison. Each individual instrument plays at a slightly different pitch, but the ensemble does not sound out of tune. The beating and phase-shift effects produced by minute tuning differences give the group a massive sound that fluctuates pleasingly over time.

Chorusing is achieved by pitch-shifting a signal up or down by a few cents and mixing the shifted signal with the original signal. You can achieve a more massive effect if you pitch-shift several copies of the signal by different amounts. One variant of this technique is to route the copies to different stereo channels. Chorus effects are often enhanced by varying the pitch shift slightly with an LFO, by a small amount of feedback, or by a bit of delay before detuning.

A *harmony processor* generates one or more harmony lines from the input signal. The simplest harmony processors transpose all input by a fixed interval. This approach, considered crude nowadays, has two drawbacks. First, it quickly bores the listener. Second (and worse), it produces notes that are out of tune. (Harmonize the first few measures of Beethoven's *Ode to Joy* in straight major thirds to get an idea of how bad this can sound.)

"Intelligent" harmony processors let you specify a scale and key, and they follow the input melody at the diatonically (or modally) correct intervals. If your music modulates a lot, you can change the scale and key parameters via MIDI. Dedicated harmony processors, such as the DigiTech Vocalist series, can perform impressive feats of multipart harmonization and pitch correction. Some high-end multi-effects units, such as the Lexicon PCM 81, also offer sophisticated pitch processing.

Varispeed imitates the sonic effect of changing the speed of a good old-fashioned analog tape deck or turntable. In the digital domain, this is easy to do—just vary the playback sample rate. When the sample rate changes, so does the pitch, producing the classic "munchkinization" effect. Given the amount of R&D that has gone into avoiding this effect in pitch/time processing, it's a bit ironic that varispeed has retained its popularity. Varispeed is always good for an audience laugh, though; you'll hear it at some point in almost any episode of *Ally McBeal*.

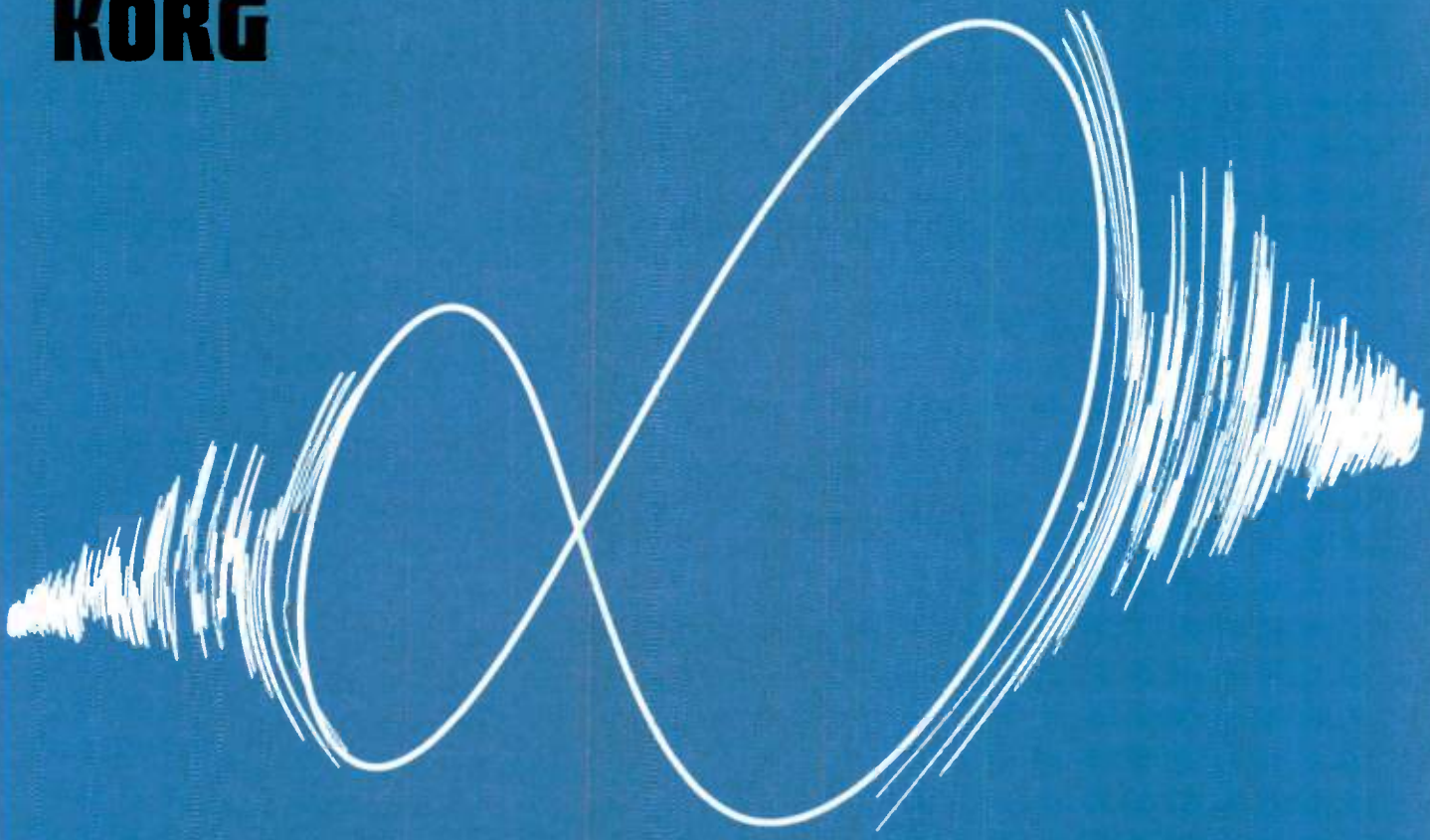
READ ALL ABOUT IT

In my articles on multi-effects processing, I've only been able to touch upon the basics of this very extensive subject. In fact, any of the effects that I've mentioned could easily be the subject of an entire article. To explore further, you might want to read through some of the excellent reference materials on the subject, such as the signal-processing sections of Curtis Roads's *Computer Music Tutorial* (MIT Press, 1996).

Multi-effects open up a huge sonic universe, and the more you know about them, the better you can put them to use in your work. Take a look at the hardware and software in your studio and see what sort of effects are lurking under the hood!

John Duesenberry's *electronic music is available through the Electronic Music Foundation. Check the EMF catalog at www.emf.org.*

KORG



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The Changing Deal

These days, the once standard record deal is anything but.

By Michael A. Aczon

The music industry is changing before our very eyes. One important facet undergoing a shift is the record deal. Long gone are the days when an unknown artist would be discovered by a cigar-smoking record executive and then catapulted to stardom with a long-term, exclusive record deal. Today's artist deals retain a few traditional contract points but have eliminated many of the elements that once were standard.

DMITRY PANICH

Topping the list of factors that have fueled such change in the industry is the artists' increased access to inexpensive yet high-quality recording equipment. This has been both good and bad news for fledgling artists. Although these tools give musicians a considerable amount of creative control over their material, major labels no longer feel the need to spend as much time and money on artist development as in the past. Consequently, large labels now concentrate their marketing and distribution efforts on artists who have "prepaid" their development dues.

Issues that would never be considered just five years ago have today become deal points in record-contract negotiations. The deals are now much more sophisticated, so artists need to focus on more than the length of the contract, the percentage of royalties, and the size of the advance. Thanks to new technology, the rise of independent labels, and a strong economy, the music business is undergoing a revolution, and the artist deal has become a major battleground.

A PLATINUM DOWNLOAD?

The advent of Internet music distribution has sent the traditional record business scrambling to adjust to the digital age. While major players in the music and computer industries hammer out new formats for the secure

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digital distribution of music, artists and record labels are heavily negotiating a number of issues concerning the digital rights to artists' masters. Because of the trend toward digital distribution in the form of downloads and streaming audio, most deals now include the exclusive rights of the labels to distribute an artist's music online.

Due to the nature of the Internet and related technology, several standard clauses in artist deals are being challenged by lawyers who need to draft contracts in response to the new technology. For instance, labels traditionally base artist royalties on the wholesale price or suggested retail price of a CD; however, because the price for downloading music from the Web has not yet been standardized, the royalties of artists who use this distribution method can fluctuate wildly.

Another label tradition—dating back to the days when music came on 78 rpm records made out of shellac—is the standard 10 percent deduction from artist royalties to pay for the many records that shattered during shipping. (Believe it or not, many of today's contracts still include *breakage* clauses.) It's difficult to justify this deduction, given digital distribution, because downloads can't break.

Labels also take a *packaging deduction* of 10 to 25 percent from artist royalties to pay for the sleeves, jackets, artwork, and jewel cases used to package records and CDs. With digital distribution, the label no longer has to pay for physical packaging, so this deduction should not apply to downloaded music.

The Internet also complicates the old practice of negotiating separate distribution deals by geographical territory. For example, an American artist who gives a French label the exclusive rights to sell an album in retail stores throughout France may find that controlling the flow of product is relatively easy. However, preventing that same label from allowing consumers outside France to download online material can be more difficult.

A related issue that is also subject to heated negotiations is ownership of Web-site domain names. Tens of thousands of dollars are being paid for a single domain name these days, making it an extremely valuable asset if an artist becomes a superstar. Labels argue that because they invest heavily in creating, developing, and marketing artists' superstar status, they should own the related domain names and Web sites. If a band establishes and maintains its own site, a record company may require that it contain links to the label's own site, which will feature not only that band but other acts signed to the label as well.

IS THAT A TURNTABLE I SEE?

On the opposite end of the high-tech discussion are *vinyl rights*. Artists who perform certain styles of music (such as techno and hip-hop) depend on underground and club play to develop their audiences. This leads to a big dilemma: vinyl is the club DJ's format of choice, but many major labels no longer press vinyl records, concentrating instead on the sale of CDs and newer formats. Therefore, artists working in these genres need to negotiate commitments from their labels to press and sell their works on vinyl.

The labels claim, however, that vinyl records are "specialty products" for target markets—promotional items that shouldn't count when calculating

artist royalties. Some artists negotiate very progressive deals that let them independently release vinyl versions of their records to reach their core audience while allowing the labels to handle distribution of "standard" CD configurations.

INSURANCE ASSURANCE

Over time, labels have discovered that recording artists are their most valuable asset. To protect that asset, most major deals include the right of the record company to take out medical and life insurance policies on its artists. Health insurance policies go into effect if, for instance, an artist should suffer from exhaustion and be unable to complete a tour or finish recording an album. Through health insurance coverage, the label can be sure that the artist will get the medical attention necessary to complete the obligations at hand. And if an artist should die before fulfilling the terms of the recording contract, a life insurance policy protects the label's share of potential royalties that the artist would have earned.

A SAMPLE PLAN

Digital sampling technology has developed to the point where sampling is not only easy but affordable. The practice of sampling previously recorded material has crossed over into so many genres that artist deals almost always include sampling liability language. When an artist wants to sample a previously recorded song, two rights must be cleared before a record including the sample can be released: the right to use the recording, and the right to use the musical composition itself.

▼

Labels and artists negotiate heatedly over the ownership of artists' Web-site domain names.

Depending on a number of factors, the cost of sample clearance can be substantial. In the evolution of the artist deal, the responsibility for clearing the samples has shifted away from the labels and onto the artist. The labels often set aside a portion of the recording budget as a sample clearance fund. This fund is used to pay any fees associated with clearing samples and, just like any other recording cost, is eventually paid back from artist royalties. The artist is assuring the label that these samples will be cleared; if problems arise in the future because of uncleared samples, any costs that the label might incur (such as paying the original label or publishing company for the sample) will be taken out of the artist's royalties.

UNBECOMING CONDUCT

"Sex, drugs, and rock 'n' roll" has been a way of life for both established and aspiring rock stars since the genre's inception. The outlandish behavior of some musicians may have been accepted in the past, but today artist deals follow the cues of society, the media, and zealous lawyers by making artists

ultimately responsible for their actions. This is accomplished with a *conduct clause*. By agreeing to this clause, an artist promises to act within the accepted behaviorial norms of society when fulfilling the duties of the contract. Artists who violate the conduct clause must reimburse the label for any costs that it might incur as a result of the misconduct. When they trash a hotel room on tour, break public-decency laws on stage, or are convicted of a violent crime, artists literally pay for their rock 'n' roll conduct through royalties.

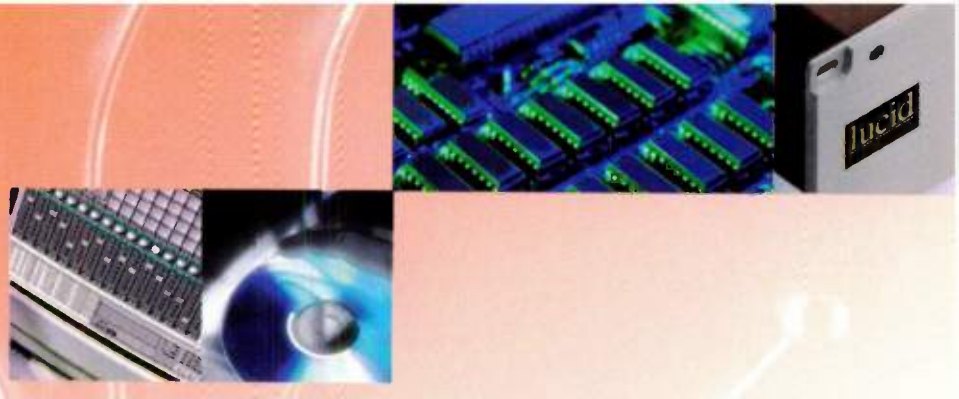
PICK A DEAL, ANY DEAL

The points discussed here so far have become integral parts of the standard label contract, but artists today have more types of deals to choose from than in the past. Using the technology industry as a model, artists now deal in "content" rather than in songs, masters, and promotional materials—treating record labels as "strategic partners" instead of gatekeepers to the record-buying public.

Major record companies are increasingly entering into nontraditional deals with independent labels. In a joint-venture deal, for example, the major and the indie co-own the venture: the major provides marketing clout and experience, and the indie provides "street credibility" and alternative marketing strategies. With a distribution deal, artists retain ownership rights to their masters and exclusive rights as recording artists, relinquishing only the distribution rights to a major label; the label therefore gets a much smaller piece of the pie than it would through a standard exclusive record deal. Other artists search outside of traditional labels and partner with online music distributors or radio stations in exchange for nontraditional compensation packages (such as 50/50 royalty splits and stock options).

Artists who are willing to stick it out through the development process—especially if they can develop sales on their own—end up with more than a few 21st-century bargaining chips to bring to the negotiating table.

When he's not surfing the Web and keeping up with new technology, **Michael A. Aczon** *can be found writing in his journal with a ballpoint pen while enjoying Little League baseball games at a park in El Cerrito, California.*



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Tips for getting the most from your gear.

Compiled by Mike Levine

In this month's installment of "Operation Help," Mark of the Unicorn helps you set up *Digital Performer* to work with *ReBirth* and ReWire, and E-mu offers tips on the Xtreme Lead-1 module's SuperBeats and Arp functions. Also, Charlie Shew of Glyph Technologies makes a guest appearance to answer a reader's question on hard drive specifications, and **EM** editor Steve Oppenheimer gives gear advice to a backpacking musician.

FIG. 1: Using ReWire technology, *ReBirth* can stream 24-bit audio directly into the mixer of Mark of the Unicorn's *Digital Performer*.

FROM THE MANUFACTURERS

Setting up Mark of the Unicorn's *Digital Performer* for *ReBirth* and ReWire

Mark of the Unicorn's *Digital Performer* is compatible with many of today's hottest plug-ins and virtual-instrument products. One is the popular *ReBirth-338*, the virtual analog drum machine/synth bass from Propellerhead Software. *ReBirth* reproduces (to the last detail) the buttons and performance characteristics of the old Roland TR-series drum and bass modules. Propellerhead (along with Steinberg) also created ReWire, a Mac OS shared library that enables *ReBirth* to stream its 24-bit audio output directly into *Digital Performer*'s virtual mixing environment in real time while both programs are running (see **Fig. 1**). ReWire even lets you control the transports from either program.

ReWire allows you to program killer drum and bass parts in *ReBirth* and then layer them with other instrument parts in *Digital Performer*'s sequencing, hard disk–recording, and mixing environment. ReWire also provides multichannel input, so you can put parts such as the kick, snare, hat, and synth bass on separate tracks in *Digital Performer* for independent automated mixing and real-time 32-bit effects processing.

To set up the ReWire library with *Digital Performer*, follow these easy steps:

1. Install *Digital Performer* (version 2.7 or later).

Will it go 'round in circles?

Fly diagonally overhead, roar by on the right, sneak up from behind or simply surround you with sonically superior performance? Yeah, it'll do that! Presenting Kind of Loud Technologies' surround production tools. Make it sound good; make it Kind of Loud.

5.1 mixing in Pro Tools® begins with SmartPan Pro, our fully automatable TDM surround panner. Featuring a proprietary circular panning paradigm (translation: a circle in which to place your sound), SmartPan Pro simplifies setting up your 5.1 mix.

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Stumped about surround? Visit www.kindofloud.com for concepts and links.

Pay attention now: If you have *anything* to do with surround, you're shortchanging yourself and your clients if you're not using RealVerb 5.1 — the first surround reverb. RealVerb 5.1 brings depth and dimension to your mix. And if you need to up-mix from stereo or mono to 5.1, there's no faster — or better — way to "repurpose" your tracks.

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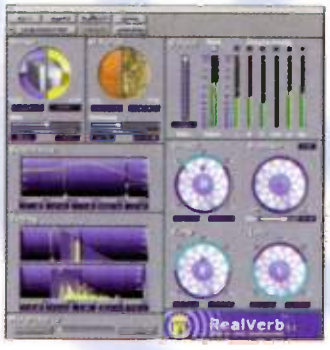

Tweetie™ and Woofie™ Use Tweetie for speaker calibration and downmix preview. Avoid unpleasant surprises with Woofie by previewing the subwoofer signal before encoding; bass redirection reroutes the sub signal to other speaker channels.

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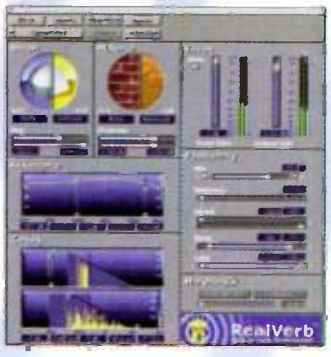
We took the core technology from RealVerb 5.1 — acoustic room simulation, room and materials blending and preset morphing — and made a great-sounding reverb plug-in at a great price.



"A dream come true. Fully automatable surround panning within Pro Tools. Sound design in a 5.1 environment with precise control over all width and movement through breakpoint editing for fine tuning and spatial creativity. Whew! SmartPan Pro is here!"

Dane A. Davis, M.P.S.E.
Supervising Sound Editor
Danetracks, Inc.

"Kind of Loud has added an entirely new, and desperately needed, dimension to Pro Tools. SmartPan Pro is now standard issue on all Pro Tools systems at Soundelux. I can't imagine doing multi-channel production without it. "

Tom Ozanich
Sound Designer/Mixer
Soundelux

"Kind of Loud is simply the only way to approach mixing in surround. SmartPan Pro makes a 5.1 mix a breeze; the depth and sonic quality of RealVerb 5.1 makes conventional reverb sound one-dimensional."

Jamey Scott
Digital Composer, Myst 3
Presto Studios

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Postal Box 3800, Santa Cruz, California 95063-3800 USA
(831) 466-3737 voice (831) 466-3775 fax info@kindofloud.com email www.kindofloud.com web

2. Use the Get Info command for *Digital Performer* and add 10 to 20 MB to its memory partition. For example, if you normally run *Digital Performer* with a 40 MB partition, try 50 or 60 MB. This adjustment reserves a little extra RAM for *ReBirth*, which will use *Digital Performer*'s memory partition.

3. Launch *Digital Performer*. Go to Basics → Edit FreeMIDI Configuration to launch *FreeMIDI Setup*. In *FreeMIDI Setup*, choose FreeMIDI Applications Only. This setup allows you to freely switch back and forth between *ReBirth* and *Digital Performer* during playback. It's almost as if *ReBirth* becomes another window within *Digital Performer*.

4. Quit *FreeMIDI Setup* and return to *Digital Performer*. Go to File → Preferences and make sure that the Play in Background option is enabled.

5. Quit *Digital Performer*.

6. Install *ReBirth-338*.

7. Launch *ReBirth* (don't launch *Digital Performer* yet). Play a few demo "mods" to make sure that it's up and running correctly.

8. Quit *ReBirth*.

9. Launch *Digital Performer*. (ReWire requires you to open *Digital Performer* before *ReBirth*.) You should now see *ReBirth* inputs in *Digital Performer*'s audio-track input menus. ReWire supplies these inputs even when *ReBirth* is not running.

10. Before you reopen *ReBirth*, either create a stereo Aux track and assign *ReBirth*'s main Mix (L/R) as its audio input, or do the same thing using a regular stereo audio track and record-enable the track. Either option activates *Digital Performer* as a ReWire host for *ReBirth*. If you don't take this step, *ReBirth* will not "see" *Digital Performer* as an audio-output destination.

Remember, you can set up a separate *Digital Performer* track for each *ReBirth* input. But you need to set up at least one track as just described.

The audio-output assignment for the *ReBirth* tracks in *Digital Performer* can be the main outs on your 2408 audio interface, the speaker outputs of your Mac, or any other available outputs.

11. Launch *ReBirth*.

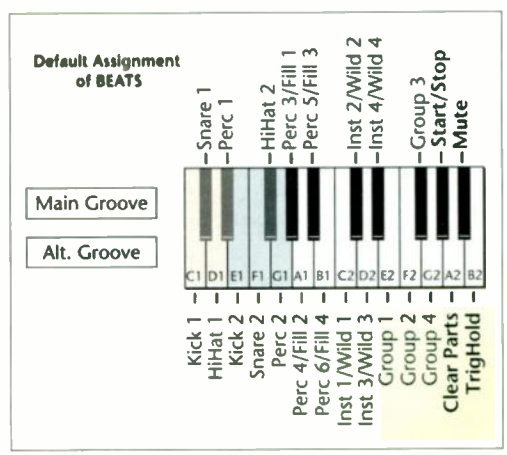
Now when you load a song in *ReBirth* and press Play in either *ReBirth* or *Digital Performer*, *ReBirth*'s audio output will appear on its assigned track on *Digital Performer*'s Mixing Board channel strip. In addition, you'll hear the audio on the *ReBirth* track's playback destination. The programs will stay in perfect sync.

Don't forget that you can apply real-time plug-in effects to the incoming *ReBirth* audio, and even bus *ReBirth* audio anywhere you want in your *Digital Performer* mix. When *Digital Performer* is in Tempo Slider mode, tempo can be adjusted in either program. In Conductor Track mode, *Digital Performer* controls the tempo, including on-the-fly tempo changes. When you're finished with your session, quit *ReBirth* first. That's it! Enjoy.—*Jim Cooper, Mark of the Unicorn*

Using the E-mu Xtreme Lead-1's SuperBeats and Arp Functions

E-mu's Xtreme Lead-1 expandable sound module now offers a SuperBeats mode, which allows you to access a built-in 16-track play-only sequencer optimized for live performance and groove creation. Based on the Beats mode introduced in E-mu's Orbit and Planet Phatt modules (which could play back only one track), XL-1 contains dozens of special 16-Part SuperBeats riffs created by some of the industry's most talented programmers.

Here's how it works. Each of the 16 Parts (instrument grooves) is assigned to a key on the selected Trigger MIDI channel. Additional key assignments give you control over the Beats: for example, Latch Notes keys turn Parts on and off; Group Parts keys trigger mul-



COURTESY E-MU SYSTEMS

FIG. 2: The E-mu Xtreme Lead-1's SuperBeats function allows the playback of up to 16 instrument grooves, which play in sync and are mapped over 2 octaves.

For manufacturer information, please see Contact Sheet, p. 205.

tiple Parts. The other options are Mute All Parts, Clear All Parts, and Start & Stop. The 16 Parts are organized into four groups—Main Groove, Alt Groove, Perc/Fills, and Inst Wild—and are mapped over two octaves (see **Fig. 2**).

The XL-1 also features the powerful Arp/Rhythm Pattern Generator, a feature that first appeared in E-mu's Audity 2000 module. Each of the pattern generator's presets offers 16 user-programmable synched arpeggiators. You can set up the XL-1 to play the arpeggiators and Beats together in a few easy steps:

1. Set MIDI mode to Omni.
2. In the Beats menu, set the Beats channel (BtsCh) to MIDI channel 01.
3. Set the Trigger channel to Basic.
4. Set the MIDI channel of the Beats preset to 01.
5. Set each Arp preset to any unused MIDI channel.

You can play only one SuperBeats preset at a time, but you can combine the Beat with up to 15 arpeggiators and other presets simultaneously (you have 16 MIDI channels to work with). And because the Arps and Beats are synched to the same Beats/Arp Clock, when you change the bpm in real time, all the Arps and Beats will automatically adjust to the new tempo. For more information about the XL-1's SuperBeats mode, check out www.emu.com.—*Derk Hagedorn and Aaron Eppolito, E-mu*

QUESTIONS FROM READERS

Hard Choices

Q: I am a music-production professional in need of a fast hard drive. I'm a little confused by the large amount of information and specifications provided by manufacturers. What are the pros and cons of high cache and high seek speed and rpm? It seems that you can have one or the other. I would like some info so I can decide what would be more beneficial for my particular applications.

Luis Gonzalez
via e-mail

A: *In order to answer your question as accurately as possible, we enlisted the help of Charlie Shew of Glyph Technologies, a company that produces high-speed, computer-based mass-storage devices for music and video production.*

First you need to understand what's happening at the drive/OS level in regard to shuttling data in and out of your application. The common wisdom on disk rotational speed (rotations per minute) is that the faster your disk spins, the better—because the less time it takes for the platter (disk surface) to come around, the faster the heads deliver your data.

Most current hard drive mechanisms come in one of three rotational-speed settings: 5,400, 7,200, or 10,000 rpm. Your choice depends on the amount of data you need to move (track count) and how many edits you will be making to that data.

For example, if you are doing radio production and you don't frequently use more than, say, six tracks recorded at 16 bits, 44.1 kHz, then you may be able to get by just fine with a 5,400 or 7,200 rpm drive. However, if you're a track junkie and you work in 24-bit, 96 kHz format with heavy edits, then you will most certainly need a 10,000 rpm drive.

As for seek time, the smaller the number, the better. The best drives offer seek times of 6 to 9 milliseconds;

at the minimum, you should get a unit with a seek time that's rated under 12 ms.

The drive cache is not as important a factor in audio work, because you want to keep data flowing steadily from the drive into your application/OS's cache. Current hard drives typically provide at least a 512K cache, and most offer 1 to 2 MB. Any of these cache sizes is sufficient for most audio work. Again, though, track junkies may want a larger drive-cache size, just to give those massive amounts of bits as much breathing room as possible as they shuttle back and forth.

The best rule of thumb is to look at the manufacturer's recommended drive specs for a particular DAW. However, the hard drive world changes very quickly, and getting information from the manufacturer on the most current drives may be difficult. Therefore, it's advisable to work with a knowledgeable dealer who knows which drive mechanisms will work best for your particular application.—*Charlie Shew, Glyph Technologies*

Trip Abroad

Q: I've recently received a fellowship that's sending me abroad for a full year to study electronic music in several countries. I'll be living out of a backpack (or two), and I need to stuff as much of a studio as possible into my luggage. I'm used to working with a sampling keyboard, a decent mixing board, an effects unit, and my laptop. The laptop is coming with me, but everything else is too big. What kind of compact gear do you recommend for the backpacking electronic musician?

Jesse Stiles
Poughkeepsie, NY

A: That depends on the kind of work you want to do. The simplest way to go is to leave the laptop at home and carry only a set of headphones (for privacy) and a small portable home keyboard with built-in speakers and a good sequencer. Some of these keyboards also have built-in computer interfaces, and almost all can be powered with batteries and AC supplies. If you're going to compose on the road and then create the finished piece at home, this might be all you need.

If you add the laptop for all the recording, synthesis, sampling, signal-processing, and mixing chores, you need a digital audio sequencer, perhaps a few DSP plug-ins, and a software synth and sampler. Make sure the programs work well together. For instance, if you use a Steinberg sequencer, use a software synth that supports ReWire technology (from Steinberg and Propellerhead Software) and VST plug-ins. Whatever your choice, think in terms of a system.

If you take the laptop, you can use any small MIDI keyboard controller because you can get your sounds from the software synth/sampler. Fatar and Novation (see **Fig. 3**) make some small, lightweight keyboard controllers that might do the trick, but again, portable home keyboards are made for this kind

TECHNICAL SUPPORT
Got a question for "Operation Help"? You can e-mail us at emeditorial@intertec.com. Published letters may be edited for space and clarity.

FIG. 3: In a lightweight traveling MIDI rig, this Novation BassStation would work well as a small keyboard controller for triggering a software synth/sampler.

of job. Get one with a built-in MIDI interface to shave off some weight.

If you plan to add vocals and acoustic instruments to your compositions when you get back, you don't have to haul around a mic and mic preamp. But if you want to sample sounds along the way (which is a cool idea) or jam with musicians you meet and record the results, a mic and preamp (a battery-powered preamp if you want to do field recording) are in the cards. Use a small, solid-state mic preamp because a tube preamp might not survive the shocks of backpacking. Similarly, dynamic mics are best for your application because they can usually take a moderate beating and still perform, and they do not require phantom power.

If your laptop has a PC Card (PCMCIA) slot, and you want to record pro-level audio, consider using an audio card such as Digigram's VXpocket (reviewed in this issue of **EM**). This card works with laptop PCs and Mac PowerBooks.

For more-specific product ideas, check out the new version of **EM**'s *Personal Studio Buyer's Guide*, available at newsstands and stores that carry **EM**. (Readers in the United States with paid subscriptions received the *PSBG* free with this issue.) The *PSBG* product listings include features, prices, and, in a few cases, dimensions, along with the manufacturers' Web URLs and e-mail addresses. You can use the *PSBG* charts to narrow the field and contact manufacturers to ascertain any sizes and weights we didn't list.—*Steve O.*

REVIEWS

DIGIDESIGN

PRO TOOLS/24 MIX AND MIXPLUS (MAC/WIN NT)

The full story on a state-of-the-art DAW.

By Jeff Burger

Digidesign is one of the pioneers in digital hard disk recording. As its products have increased in sophistication, it has gone from merely replacing the multitrack tape deck to offering the promise of a complete computer-based recording studio.

Does the hardware/software package of Pro Tools/24 Mix and *Pro Tools* 5.0 fulfill that promise? Before I answer that question, allow me to set the stage for a moment with an overview of Pro Tools and the 24 Mix system.

WHY PRO TOOLS?

More companies than ever are making digital audio workstations that promise to transform your computer into a desktop studio. Most of these products are host based, which means your CPU does most or all of the processing and disk I/O. While a host-based system lets you handle sessions that incorporate lots of tracks and effects, you will have less flexibility when it comes to real-time processing on larger projects. To get around this, you will need to perform file-based processing.

Several other factors become important in host-based environments, including Digidesign's own Digi 001. First, getting signals in and out of a computer involves an inherent latency (the delay that occurs when you're recording or playing through the system). Under

Together, Digidesign's Pro Tools/24 Mix and *Pro Tools* 5.0 are a powerful cross-platform digital audio workstation/sequencer that gives you up to 64 audio tracks and 128 MIDI tracks.

THE EVOLUTION OF SOUND CONTINUES

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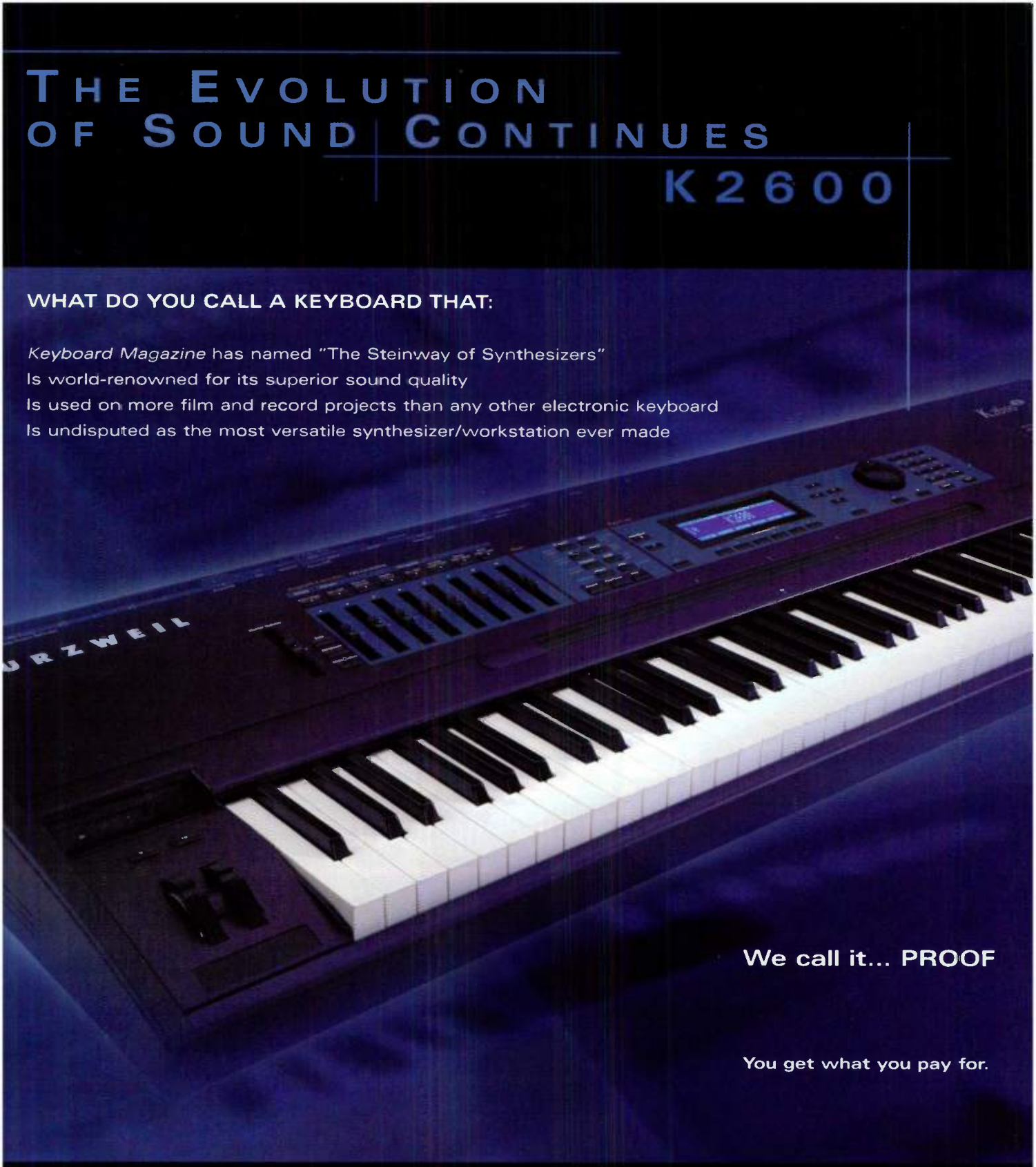

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Accessorizing Pro Tools

One of the most attractive elements of the Pro Tools system is the wide variety of hardware and software accessories available from both Digidesign and third-party developers (referred to by Digidesign as *development partners*). These additions allow you to expand and customize the system in an amazing variety of ways.

However, documenting all of them would require a book, so I have covered the subject according to the needs of the personal studio. Further information on many of the products in this section can be found on Digidesign's Web site (www.digidesign.com).

PLUG-IN UNIVERSE

There's a TDM plug-in out there to emulate just about anything you can do with hardware—and to achieve effects that come only by way of computer processing. Interestingly, most TDM plug-ins ship with file-based AudioSuite corollaries. This means you have a choice of either real-time or file-based processing.

Remember that there is more to learn about each plug-in than simply how it sounds. Although the subject is beyond the scope of this article, you will gain a clearer understanding of each plug-in you work with if you ask yourself these questions: How much DSP power does it require? Does it take an entire chip? How many instances can you get per chip? Does it require SRAM or DRAM?

Dynamics processors. Any engineer will tell you that every dynamics processor has its own personality. *Drawmer Dynamics* (Mac/Win NT; $595) comes with a gate/compressor/limiter based on Drawmer's DS201 and an expander/compressor/limiter based on Drawmer's DL241 and DL251. Both are mono, the former designed for highly transient material like percussion, the latter for vocals and less dynamic instruments.

***The Shuffler*, from the GRM Tools, vol. 1, bundle, randomly slices an audio file into small parts and reassembles it according to user definitions.**

Similarly, the Focusrite *d3* (Mac/Win NT; $795) is modeled on the popular Focusrite Red Range 3 Dual Compressor/Limiter. Known for being very forgiving (set the knobs anywhere and it sounds great), this plug-in is very true to the hardware version.

Bomb Factory recently introduced its *Classic Compressors* (Mac/Win NT; $599), meticulous models of the omnipresent Teletronix LA-2A and Urei 1176, with the digital addition of a sidechain input. Bomb Factory's plug-ins all have phenomenal sound quality and are a dead-on emulation of these classic processors.

McDSP's *CompressorBank* (Mac; $495) is a great plug-in that can be set to model just about any dynamics hardware. Presets include the dbx 165, the Neve 33609C, the LA-2A, and the 1176 LN.

Waves' *C4* (Mac/Win NT; $595) and Antares *MDT* (Mac; $249) are multiband dynamics processors—4 and 5 bands, respectively—that provide optimum control compared with single-band units. The C4 is the only processor with parametric dynamic control, allowing you to set different response slopes for different frequency ranges—there's no hardware unit that does this. This is also one of the best compressors I've heard.

In the limiting department, Digidesign's *Maxim* (Mac/Win NT; $445) goes beyond what analog limiters can do by looking ahead in an audio file to anticipate and reduce peaks. *Maxim* gives you the ability to increase the gain by as much as 12 dB and still retain harmonic detail.

Tone-sculpting. Classic EQs have their own character as well. Focusrite's *d2* (Mac/Win NT; $995) models the Rupert Neve–designed Red Range 2 Dual EQ, and allows you to use up to 6 bands.

McDSP's *FilterBank* (Mac; $495) sounds great, provides loads of flexibility, and comes with enough presets to fill just about any EQ need. Both *d2* and *FilterBank* can be run in mono or stereo.

Aphex's *Aural Exciter* (Mac; $495) adds the same harmonic enrichment found on tons of hit vocal tracks to your TDM tool chest. Similarly, Aphex's *Big Bottom* (Mac/Win NT; $345) uses the principles of the Aphex Model 104 to add low-end punch and presence without boosting peak level.

Reverb. If you're a fan of the Lexicon sound, *LexiVerb* (Mac; $795) gives you the company's chamber, plate, inverse, and gate algorithms with a 3-D graphic interface and 100 presets. If you really want control, Kind of Loud's *RealVerb* (Mac; $695) lets you design your own room including shape, size, and materials. It even lets you morph between presets.

***Pitch 'n Time* by Serato Audio Research lets you perform extreme time-compression and pitch-shift effects with few audible artifacts.**

Sound generation. While the polyphony of most other soft synths is still dependent on the host CPU, Digidesign's DSP-based *Access Virus TDM* (Mac/Win NT; $795) isn't. *Access Virus TDM* gives you 16 voices and 8 timbres per chip while bringing all of the techno sounds and tempo-based triggering found in the hardware version.

Having a problem with a less-than-perfect drum track? Digidesign's *Sound Replacer* (Mac/Win NT; AudioSuite only; $395) replaces drum hits and other sounds containing high transient content with files from a sample library on your hard drive.

Modeling miscellany. Can't afford a chest of mics or a roomful of guitar amps? Antares' *Microphone Modeler* (Mac/Win NT; $599) takes the signal from your source mic and models a variety of world-class transducers, while allowing you to vary the amount of tube warmth. Line 6's *Amp Farm* (Mac/Win NT; $595) and Bomb Factory's *Sans Amp PS-1* (Mac/Win NT; $499) are software versions of the Line 6 Pod and Sans Amp PS-1, respectively.

If you're looking for a virtual Leslie, check out Bomb Factory's *Voce Spin* and *Voce*

Chorus/Vibrato (Mac/Win NT; $395). The company also collaborated with Bob Moog for a TDM version of a Moogerfooger filter and ring modulator combo (Mac/Win NT; $399).

Special effects. Whether you're bored or just looking for a new sound, special effects will keep you busy for quite a while. Digidesign's *D-Fi* (Mac; $395) includes different plug-ins for adding lo-fi, sci-fi, subharmonics, and other effects to your tracks. Digidesign's *Bruno/Reso* combo (Mac/Win NT; $395), among other things, takes vocoding to the next level. This plug-in uses cross-synthesis techniques to transform source material into unique sonic textures—including tasty rhythmic effects from keyed inputs.

Give a bit of DSP to some of the engineers responsible for the Eventide Ultra-Harmonizer, and you get Wave Mechanics' outrageous *SoundBlender* (Mac/Win NT; $495). *SoundBlender* combines pitch, filter, and modulation effects in one plug-in. The Modulation Matrix and tempo control open the processing possibilities even further.

Pitch- and time-processing. Less-than-perfect vocals? Wave Mechanics' *PitchDoctor* (Mac/Win NT; $495) corrects intonation in real time and gives you the option of preserving formants to avoid the dreaded "chipmunk" effect. Antares *Auto-Tune* (Mac; $599) performs similar functions and lets you specify a wide variety of popular or user-defined scales. *Auto-Tune* can correct even the most tone-deaf performance.

If you want to change the key or octave of a performance, check out Digidesign's *DPP-1* (Mac/Win NT; $495) for a four-octave range. Wave Mechanics' *PurePitch* (Mac/Win NT; $495) gives you six octaves complete with formant control over male and female vocal quality.

If you're looking for a high-quality tool to fit your tracks to a given tempo, pitch, or length, you'll need file-based processing: Synchro Arts' (www.synchroarts.co.uk) *ToolBelt* (Mac; stand-alone; $425) is the baseline for high-quality processing; Serato's *Pitch 'n Time* (Mac; AudioSuite only; $799) and Wave Mechanics' *Speed* (Mac/Win NT; AudioSuite only; $495) have additional tricks for handling dense harmonics. Both perform phenomenally during extreme changes in pitch or time.

***Filterbank,* by McDSP, allows you to create your own stereo or mono EQ plug-in using up to six bands.**

Spatial tools. In addition to fattening thin tracks and creating complex multitap delays, *WideR* (Mac; $249) from Cuan (www.cuan.com) can create virtual 3-D audio from your stereo mixes. If you're looking to get into surround sound, *Dolby Surround Tools* (Mac; $795) lets you preview the results in a 4-2-4 monitoring environment before you send your tracks to be matrix-encoded for theater audio.

Kind of Loud's products allow you to realize finished 5.1 mixes. With *SmartPan Pro* (Mac; $995) you use a graphic interface and an optional joystick to create surround mixes. *RealVerb* 5.1 (Mac; $1,495) is the only plug-in that handles multichannel reverb. *Tweetie* (Mac; $395) aids in the calibration of speakers for surround mixing, and *Woofie* (Mac; $395) helps you accurately preview the subwoofer's signal even without the real thing.

Noise reduction. Need to rid your tracks of air conditioner hum or guitar-amp buzz? Try Digidesign's *DINR Intelligent Noise Reduction* plug-in (Mac/Win NT; $995). If you need point-and-shoot satisfaction in cleaning up old vinyl and tape recordings, Arboretum's *Ray Gun* (Mac/Win NT; AudioSuite only; $99) is a great weapon. Arboretum's big gun, *Ionizer* (Mac; AudioSuite only; $499), adds 512-band EQ, pseudo vocoding, and the ability to apply the frequency fingerprint of one sound onto another.

***SmartPan Pro,* by Kind of Loud Technologies, allows you to precisely position audio elements within an LCRS, 5.1, or 7.1 surround mix.**

Mastering and metering. You don't have to leave Pro Tools to create your final master. The folks at TC Works, creators of the Finalizer mastering processor, have put their experience into *Master X* (Mac; $995), which includes a 3-band expander/compressor/limiter with dithering. *MasterTools* (Mac; $495) utilizes Apogee's world-class UV-22 dithering algorithm to provide excellent bit reduction during mastering. *I-Media Audio MPEG-2* (Mac; AudioSuite only; $495) from QDesign gives you full control over the stereo compression of your files for DVD mastering and other purposes.

If you really want to get tweaky, Metric Halo's *SpectraFoo* (Mac; $500) provides state-of-the-art metering, scopes, spectrum analysis, and more for critical engineering work. *SpectraFoo Complete* (Mac; $1,000) adds various code-metering and a 24-bit signal generator.

Combination plug-ins. With DSP seemingly always at a premium, these products offer multiple functions in a single plug-in while putting all the controls in one place. Waves' *AudioTrack* (Mac/Win NT; $150) offers a 3-band EQ, a compressor, and reverb in a single tool. Antares' *JVP* (Mac; $199) combines compression, expansion, gating, de-essing, parametric EQ, and delay effects under one roof. TC Works *VoiceTools* (Mac; $699) integrates pitch correction with a compressor, a de-esser, a 3-band EQ, saturation, and gate.

Metric Halo's *ChannelStrip* (Mac/Win NT; $999) goes beyond the average mixer channel to provide gain/trim, polarity inversion, expander/gate, a compressor, a 6-band 48-bit parametric EQ, selectable filter types, and metering. Its sibling, *ChannelStrip/SP* (Mac/Win NT; $495), offers much of the functionality to those on slimmer budgets.

Bundles. Some of the best deals are the plug-in bundles. Perhaps the best-known bundles are from the Israeli company Waves. The TDM Bundle (Mac/Win NT; $1,000) includes the popular *L1* limiter, the *C1* compressor/gate,

Accessorizing Pro Tools

the *Q10* paragraphic equalizer, the *TrueVerb* reverb, the *PAZ* psychoacoustic analyzer, a stereo-image enhancer, and more. In addition to a bass maximizer, a de-esser, and a pseudostereo processor, the TDM II Bundle (Mac/Win NT; $1,000) includes the classic *Renaissance Compressor* and *Renaissance Equalizer*.

Pro-FX Plus (Mac/Win NT; $700), the latest Waves bundle, includes a flanger, AM/FM/panning modulation effects, a harmony generator, multitap delay, a Doppler simulator, and a unique creative-filtering plug-in called *Enigma*. The top of the line, however, is the Gold TDM Bundle (Mac/Win NT; $700), which combines 17 Waves plug-ins for the Mac and 16 for the PC. Waves bundles offer the most bang for the buck.

The Power Looping generator in Synchro Arts' *ToolBelt* gives you control over a number of important parameters as well as a view of the waveform.

Spain's DUY offers the amazing (and somewhat underrated) DUY TDM Tools bundle (Mac; $1,295). This collection includes a classic-tube modeler, a 3-band frequency shaper, a multichannel spatial enhancer, a level maximizer, and a tape simulator. On the other hand, one of the most incredible plug-ins out there is the DUY *DSPider* (Mac; $1,199). *DSPider* is a plug-in construction kit consisting of 40 modules that you can assemble in almost any configuration. The are also over 220 presets to choose from.

France's INA-GRM offers GRM Tools, vol. 1 (Mac; $349), with four plug-ins: a 23-band EQ, Doppler simulation, five comb-filter variations, and a shuffler that randomly slices and reorders microscopic bits of audio. GRM Tools, vol. 2 (Mac; $349), includes four more: a band-pass filter, 24-tap delay, dual pitch-shifters, and *Freeze*, a plug-in that extends a sound by the desired amount via looping.

Finally, Germany's TC Works offers *TC Tools* (Mac; $999), which includes exceptional-sounding chorus/delay, EQ, and reverb.

DIRECTCONNECT

Digidesign's new *DirectConnect* plug-in allows you to take virtual instruments and other sound sources right into the TDM mixer. Digidesign is still working on a Windows NT version as this goes to press, so I'm covering only Mac products here.

Koblo Studio9000 ($595) provides two different analog synthesizers, a polyphonic sample-based synth, and a drum module in a single host-based package. The sound quality is great, and the interface is easy on the eyes.

Reaktor ($499), from Native Instruments (www.nativeinstruments.com), puts the full functionality of a modular synth in your computer. Its *Dynamo* ($199) is a collection of 25 sound machines for creating unusual soundscapes.

BitHeadz's (www.bitheadz.com) popular *Retro AS-1* ($259) creates analog synth sounds, both vintage and new. BitHeadz's *Unity DS-1* ($449) brings a powerful sampler into Pro Tools as well. Both engines take the form of system extensions in an effort to yield as much polyphony as possible. *Unity* reads both native and Akai formats. Other formats can be converted using BitHeadz's *Osmosis*.

While Pro Tools alone isn't a looper's dream, *Alkali* (Mac; $299) from Audio Genetics (www.audio-genetics.com) works with REX loop files to instantly fit loops to any tempo without artifacts.

HARDWARE

Control surfaces. One of the few complaints people have about Pro Tools and similar DAWs is the lack of tactile control. However, you can change that with control surfaces that put physical faders and more under your fingertips. Most models allow the fixed number of physical faders to address multiple banks of virtual tracks. In certain cases, you can chain together multiple units. All work on either Mac or Windows NT systems.

Given some patience, you can get just about any MIDI controller to talk to Pro Tools via SysEx. Pro Tools ships with preconfigured profiles for several control surfaces. The CS-102 ($795) from JLCooper (www.jlcooper.com) offers eight 100-millimeter faders, six knobs, an optically encoded jog/shuttle wheel, and more. Its MCS-3800 ($2,999) has eight moving faders, five knobs, a VTR-style jog wheel with shuttle ring, an LCD, a numeric keypad, and options for controlling other hardware such as video gear. The PC1600x ($399) from Peavey (www.peavey.com) offers 16 faders, a jog wheel, and an LCD display, and it's a great bargain.

Unlike the aforementioned products, the Mackie (www.mackie.com) HUI ($3,499) was designed specifically to control Pro Tools, including eight motorized faders, a jog wheel, a numeric keypad, LED meters, an analog monitor section with two mic preamps, direct controls for many Pro Tools functions, and much more.

You can get some of the same power—including the least-expensive moving faders on a controller—with CM Automation's MotorMix ($999) which makes use of the HUI profile. At the top of the list of controllers is Digidesign's dedicated ProControl ($11,995). ProControl gives you access to every externally controllable Pro Tools function. And, all communication is handled via ethernet to keep your MIDI channels unburdened.

Expansion slots. If you've run out of slots for Mix cards, you can explore the expansion-chassis options. SBS Expansion Units (www.sbs-cp.com) makes rack-mount chassis that provide 7 or 13 slots (Mac, $1,596; Win NT, $1,846). Magma (www.magma.com) also makes 7- and 13-slot units (Mac, $1,495; Win NT, $1,995) that can be used with Apple G3 PowerBooks.

No-muss Ultra Wide SCSI acceleration can be found in the dual-channel SCSI64 Kit (Mac/Win NT; $545), which includes two cables. And while Pro Tools isn't as picky about drives as it once was, Digidesign's 9 GB and 18 GB DigiDrives (Mac, $1,380; Win NT, $2,200) are as good as it gets when it comes to compatibility.

Synchronization and video. Digidesign manufactures several hardware options for integrating Pro Tools with other media. The Video Slave Driver (Mac/Win NT; $995) works with a SMPTE-to-MIDI converter to sync Pro Tools to black burst or house sync. Add SMPTE slaving and a VITC reader/generator with window burn, and you have the Universal Slave Driver (Mac/Win NT; $2,095), for synchronization with an accuracy of ±1 sample for free-running SMPTE.

"In the future, hundreds of microphones will fit in a single rack space."

*> H.D. Wells, Bromley, England, 1899**

Thanks to the new AMM-1 Microphone Modeler from Antares, the future is now.

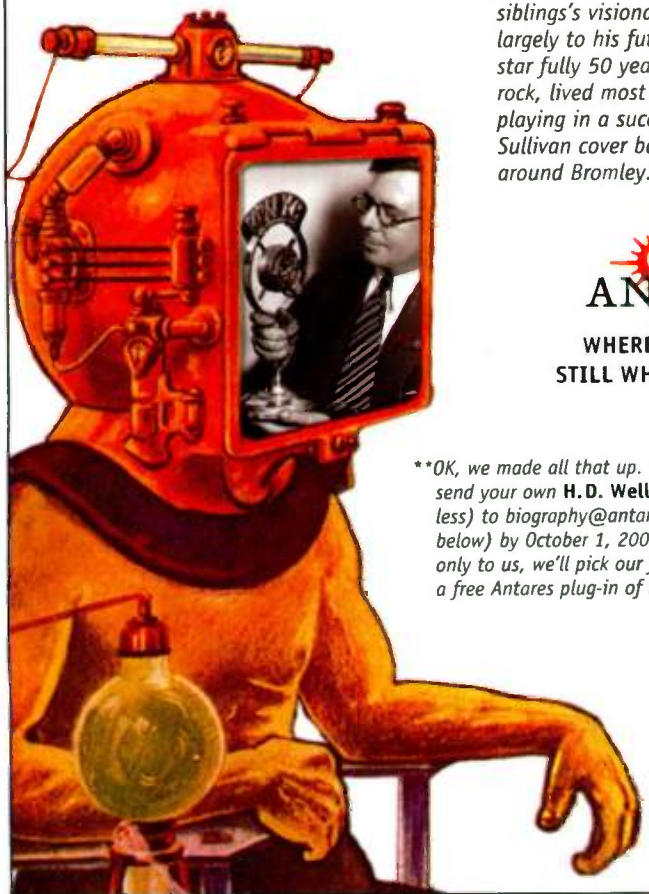
Inside this unassuming (but stylish) 1U rack is contained the spirit of many of the world's most revered microphones. Based on the same technology that made our Antares Microphone Modeler the most talked-about software plug-in of 1999, the AMM-1 lets most any reasonable quality microphone sound like any of a wide variety of high-end studio mics.

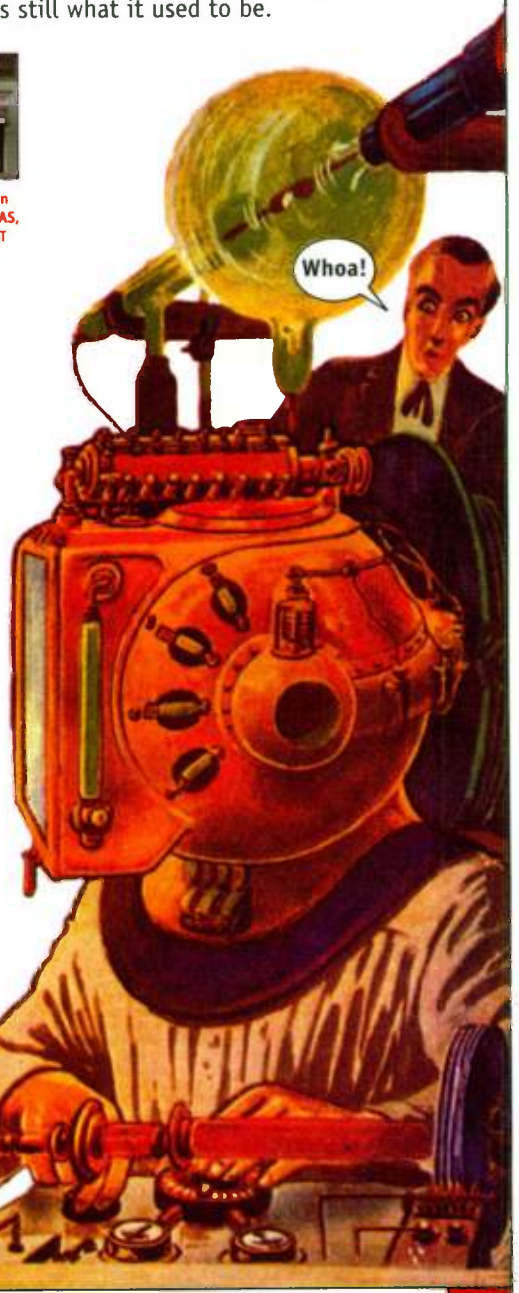
From historical classics to modern exotics to a selection of industry-standard workhorses, simply tell the AMM-1 what microphone you are actually using and what microphone you'd like it to sound like. It's really as simple as that. You can even mix and match the bass and treble characteristics of different microphones or add the warmth of classic tube saturation.

And with over 100 mic models built in, as well as the ability to add new models downloaded from our web site, the AMM-1 will insure that you always have the right mic sound for every track.

So check out the new AMM-1 at your local Antares dealer. Where the future is still what it used to be.

And don't forget the plug-in version available for TDM, MAS, RTAS, DIRECTX and MAC VST





H.D. Wells, *H. G. Wells's little-known older brother, shared his more famous siblings's visionary acumen but, due largely to his futile desire to be a rock star fully 50 years before the arrival of rock, lived most of his life in obscurity, playing in a succession of Gilbert & Sullivan cover bands in pubs in and around Bromley.***

ANTARES

WHERE THE FUTURE'S STILL WHAT IT USED TO BE

***OK, we made all that up. Think you can do better? Then send your own* H.D. Wells Biography *(in 100 words or less) to biography@antarestech.com (or to the address below) by October 1, 2000. Using arcane criteria known only to us, we'll pick our favorite and send the author a free Antares plug-in of his or her choice. Really.*

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uses another 40 MB of memory for hard drive buffering. With an operating system that requires 20 MB (Pro Tools/24 Mix requires Mac OS 8.6 or greater), your minimum system requirement is 128 MB of RAM. You'll need to allocate another 20 MB to DAE if you plan to run a lot of plug-ins or use other applications simultaneously.

Because a Pro Tools/24 Mix system doesn't rely much on the CPU, you can get away with running it on older computers, such as a Mac 9500/150 MHz. (As a matter of fact, 9500s and 9600s have maintained their market value because they are the last Macs to have six PCI card slots.) I currently run a Pro Tools/24 Mixplus system on a beige G3/300 MHz, with two of the three slots filled with Digidesign cards. Since my machine has video and SCSI support on the motherboard, I still have a slot free for a SCSI accelerator if necessary.

The Mac G4 is another story. Because it doesn't have a built-in SCSI port, you'll be forced to devote a slot to a SCSI card. (Digidesign says that you can run small sessions using an internal IDE drive, but the company doesn't recommend it.) If you need additional card slots, expansion chassis are available from third parties (see the sidebar "Accessorizing Pro Tools").

Your choices are more limited on the PC side of things. The bittersweet news is that TDM-based Pro Tools runs only under Windows NT (specifically Workstation 4, Service Pack 5). The NT version requires 192 MB of RAM to record 32 tracks, and 256 MB of RAM for 64 tracks.

FIG. 3: The ADAT Bridge I/O gives you a number of interface options, including S/PDIF, AES/EBU, Lightpipe, and slave clock I/O. Analog and word clock outputs are also included.

However, due to the complexities of the PC world, Digidesign has certified only a single non-Mac machine for use with Pro Tools—the IBM IntelliStation M Pro. That's not to say that it won't run on other CPUs, but you're basically on your own—especially when it comes to SCSI implementation. The IBM IntelliStation M Pro gives you five free PCI slots for Digidesign hardware.

Although the IntelliStation M Pro has more slots and potentially more power than a Mac, it costs around $4,000. By comparison, you can get a decent Mac for about $1,600. The Pro Tools hardware itself is identical for Macs and PCs, so you can migrate your investment if you need to.

It's interesting to note that much of the music and post-production world is Mac-based while the corporate and broadcast crowd gravitate toward NT. And, there are more add-ons, such as third-party software, available for the Mac platform.

DRIVES AND SCSI

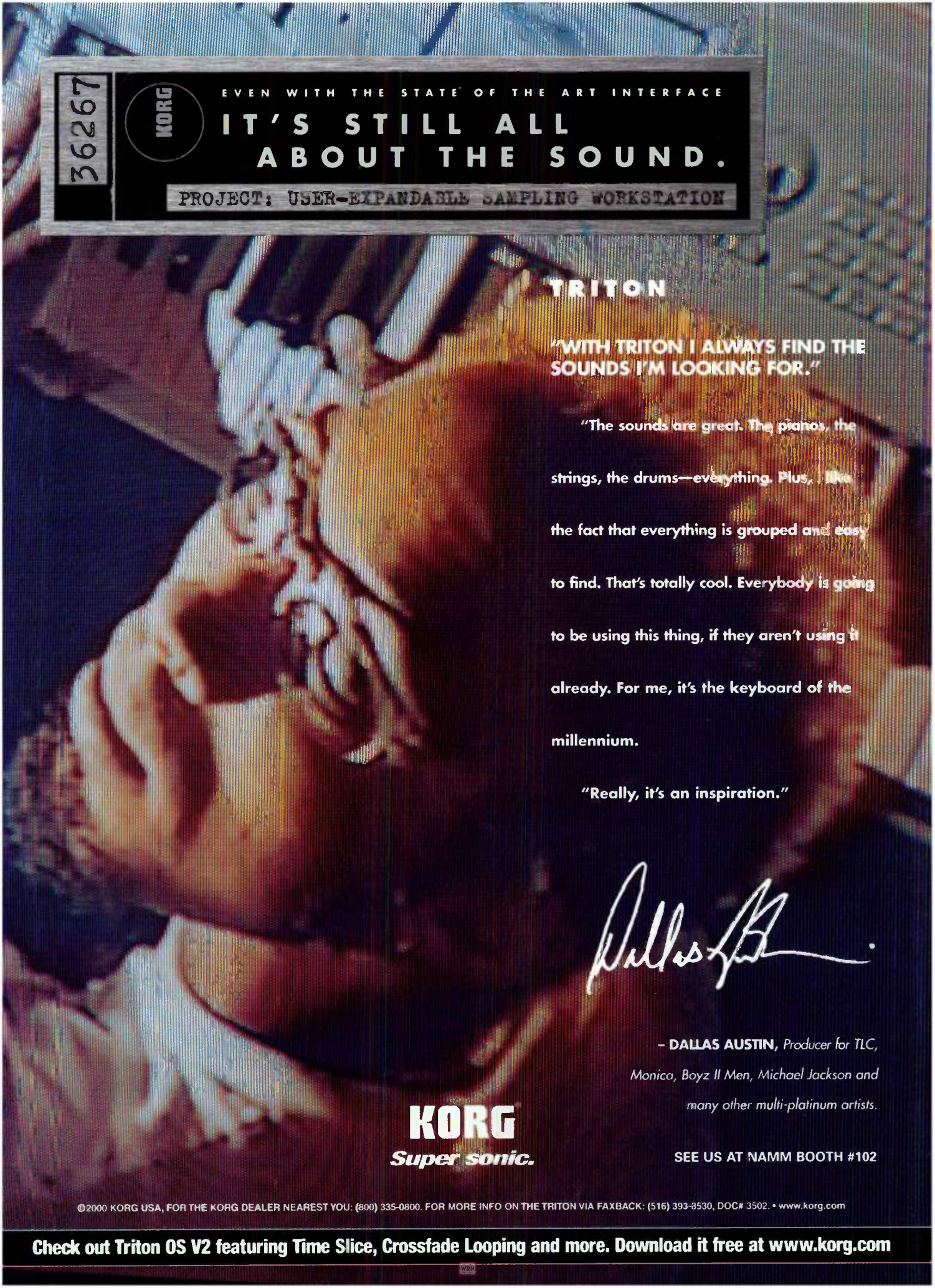
Regardless of platform, the days of connecting dedicated audio drives directly to Digidesign PCI cards are long gone. The ability to use standard SCSI ports (internal or external) makes drive management easier and more flexible than with older Pro Tools systems. In theory, you can even record to the drive where your system software resides. However, this is not recommended, in part because audio drives should be defragmented regularly: the hidden authorizations of music applications and plug-ins on the system drive make that a nightmare.

Similarly, the speed of today's hard disks means that there's no longer a need to adhere strictly to Digidesign-certified drives—although by doing so, you can be confident that the drive you purchase will perform up to Digidesign's standards. The company tests drives using a 32-track session with three edits per second and plenty of crossfades. Theoretically, if a drive can keep up with that unlikely scenario, it should be able to handle most anything.

Pro Tools/24 Mix and Mixplus Interface Specifications

Interface	Analog I/O	Digital I/O	Digital I/O Resolution	A/D and D/A Conversion	Metering per Channel
888/24 I/O	8 in/out (XLR)	8 AES/EBU in/out; S/PDIF in/out	24-bit	24-bit	15-stage ladders (output only)
888/20 I/O	8 in/out (TRS)	S/PDIF in/out	24-bit	20-bit	single-activity LEDs (output only)
888 I/O	8 in/out (XLR)	8 AES/EBU in/out; S/PDIF in/out	24-bit	18-bit	15-stage ladders (output only)
882 I/O	8 in/out (TRS)	S/PDIF in/out	24-bit	18-bit	single-activity LEDs (output only)
1622 I/O	16 in/2 out (TRS)	S/PDIF in/out	24-bit	20-bit	3 LEDs for each in and out
ADAT Bridge I/O	2 outs (TRS)	2 Lightpipe I/O; AES/EBU; S/PDIF	20-bit, 24-bit	20-bit out	status LEDs

For manufacturer information, please see Contact Sheet, p. 205.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The point at which your system needs a SCSI accelerator card varies because SCSI performance is influenced by session bit depth, edit density, and track count. As a rule of thumb, you can run 16 to 24 tracks with a low edit density before you'll need an accelerator. If you spring for an Ultra-Wide accelerator, it's almost guaranteed that it will keep up with any drive you purchase in the near future.

Storage requirements depend a great deal on the bit rate: 24-bit sessions require more storage space, obviously, than 16-bit sessions. In a 24-bit session, you get about 130 minutes of track time for each gigabyte of storage, while a 16-bit session yields about 200 track minutes for each gigabyte. A 9 GB drive holds about 1,100 track minutes of 24-bit audio, or about 45 minutes of a 24-track session. And the arithmetic scales linearly: an 18 GB drives yields 2,200 track minutes of 24-bit audio, or 90 minutes of a 24-track session.

MIX IT UP

Pro Tools' Mix Farm cards are identical to Mix Core cards, with one exception: core cards have a special ID chip that *Pro Tools* needs to see in order to run. This means you can't short-cut the system price by purchasing just a Farm card.

Both Mix PCI cards attach to the hardware interface with a 50-pin connector via a 12-foot cable. Each card can handle 16 audio inputs and outputs using either a single 16-channel interface or two 8-channel interfaces connected with a Y-cable. You can mix and match interfaces in a system, up to a maximum of 72 hardware inputs and outputs. You can also use inputs to bring line-level instruments (such as synthesizers, samplers, and other gear) into a mix. The limit for real-time mixing is 64 simultaneous digital audio tracks and 64 live inputs, for a total of 128 channel strips.

The Mix cards also include a 9-pin serial-port connector, which you can use to connect Digidesign's Universal Slave Driver or an external controller. Because older Pro Tools systems usurped modem or printer ports for this purpose, the 9-pin connector is a welcome addition.

GOOD HOUSEKEEPING

When calculating DSP power, what is commonly overlooked is that mixing requires DSP as well. Specifically, you can run up to 26 channel strips from one chip.

As mentioned earlier, Mix Core and Mix Farm cards each have six Onyx chips. Three types of Onyx chip RAM configurations are on each card, and the way the system uses a chip depends on the presence and type of RAM on the chip. For example, each card has one chip without RAM that is used for mixing, EQ, compression, and other non-time-based processes. The two DRAM chips on the card are typically used for time-based effects, such as delays. Also on the card are three SRAM chips, which are optimal for RAM-intensive time-based effects like reverbs.

Pro Tools sessions larger than 32 tracks on a single PCI-card Mix system use two chips for mixing—a non-RAM chip and an SRAM chip. In a Mixplus system, the non-RAM chip on each card is assigned to mixing tasks, leaving all the DRAM and SRAM chips in the system free for effects processing.

In addition to the upgraded DSP complement, Pro Tools/24 Mix systems provide for DSP sharing by way of Digidesign's MultiShell technology. In previous versions of Pro Tools, the first instance of a given TDM plug-in—say, a compressor—took over a DSP chip. You might be able to get a number of instances of the compressor from the same chip, but the chip was otherwise unavailable to additional plug-ins, even if it had unused processing power in reserve.

Digidesign's new DSP Manager dynamically allocates the processing power of the Onyx chips as MultiShell plug-ins are added to a session. If several instances of a MultiShell plug-in use only part of a chip, the remainder of the chip's processing power is available to other MultiShell plug-ins when a new TDM plug-in is invoked. Together with DSP Manager, a maximum of five MultiShell TDM plug-ins can share the same DSP chip at once. Developers can require that their plug-ins be loaded on a specific type of chip or merely given a preference for a certain chip type.

For example, consider a session in which a reverb plug-in that prefers (but does not require) an SRAM chip is allocated to one. If you add another reverb plug-in that explicitly requires SRAM, the DSP Manager shuffles the original reverb to a DRAM chip to make way for the higher-priority reverb plug-in.

Pro Tools 5.0 allows you to monitor DSP usage and other system resources in real time. If you find yourself regularly hitting the DSP ceiling in 24-bit sessions, running a 16-bit session will, in most cases, allow you to run more plug-ins. The sound quality of the 16-bit session won't be as high as that of 24 bits, but it will be far from shabby.

INTERFACES PRIMER

As I mentioned earlier, the prices of these Mix systems do not include the hardware audio interface. Although an interface adds to the cost of a system, the number of I/O options available gives you a large degree of flexibility in customizing your setup.

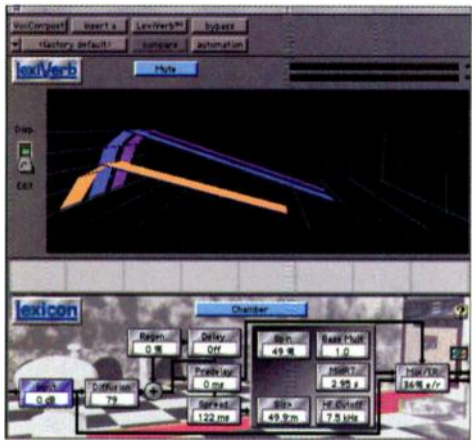
The Pro Tools/24 Mix system is also backward compatible. For example, I was able to use the 882 I/O from an old Digidesign Session 8 system as the interface for the Pro Tools/24 Mix system.

All of Digidesign's interfaces include a pair of BNC connectors that provide Superclock word-clock I/O for synchronization with other interfaces or digital devices. Superclock runs at 256 times the sample rate, which accommodates TDM's 256 pathways.

Digidesign's hardware interfaces will

Pro Tools Specifications

Audio Tracks	64
MIDI Tracks	128
Buses	32
Sends per Track	5
Inserts per Track	5
Resolution	24-bit, 16-bit
Sequencer Tracks	128 maximum
Sequencer Resolution	960,000 ppqn, synched to internal or MIDI Clock source
Quantization	whole note to 64th-note triplet



***LexiVerb* models four classic Lexicon reverb algorithms and allows you to control groups of parameters with user-defined macros.**

automatically slave to a valid clock when in internal sync mode. The interfaces can also sync to an external clock when a signal is present at the first two channels of digital audio inputs. While all Digidesign inputs have status lights for sync mode, sample rate, and input mode, the switching of settings is done from the software.

The interfaces handle both 44.1 and 48 kHz sampling rates. Digidesign has no immediate plans to implement 88.2 and 96 kHz sampling capabilities.

THIS YEAR'S MODEL

The interface you choose will depend on both your studio setup and the way you like to work. If you prefer working on a traditional mixing console, you may want an interface with a large number of outputs. If you enjoy tackling complex mixes on the computer, and you have the RAM and cards to do it, a pair of outputs for making 2-track masters may be all you need. However, the more inputs and outputs you have, the more connectivity you have with the outside world.

888/24 I/O. The flagship of Digidesign's interface line is the 888/24 I/O ($3,695). On the front panel of the 888/24 I/O (see **Fig. 1**) are 15-segment LED ladders, which display the output level for each channel. (The input levels are displayed onscreen.) However, the main action is inside the interface and on the rear panel.

The 888/24 I/O offers eight analog inputs and outputs on XLR connectors and 24-bit converters. While the unit is factory set to +4 dBV, you can switch each port individually to −10 dBu by removing the cover and changing the internal jumpers. It also ships from the factory with the analog ins and outs calibrated at a nominal level of −18 dB, thus providing 18 dB of headroom. A front-panel trim pot for each input and output allows you to calibrate the levels.

The 2U device also offers eight channels of AES/EBU I/O on XLR jacks and two channels of S/PDIF I/O on RCA jacks. The active inputs are selected in pairs via software. All three output formats (one analog and two digital) are always live, allowing you to monitor the analog outputs while sending digital signals to digital devices. The unit also acts as an 8-channel A/D/A converter when your computer is off or disconnected.

As you might expect at this price, the sound of the 888/24 I/O is excellent. Discussions of whether the unit's converters sound better than high-end third-party converters boil down to a

matter of taste. There's a small amount of fan noise in the Digidesign unit, but it will be masked in many rack situations.

882/20 I/O. The 1U 882/20 I/O ($1,245) is a 20-bit device with 8 inputs and 8 outputs on balanced TRS jacks (see **Fig. 2**). The input levels are switchable between +4 dBV and –10 dBu as a bank rather than individually. The same applies to output levels. Factory calibration is at –14 dB, for 14 dB of headroom.

The 882/20 I/O can function as a stand-alone 2-channel A/D/A converter, but it sends and accepts only S/PDIF digital signals. LEDs for each channel indicate the presence of signal above –30 dB. Like the 888/24 I/O, inputs are monitored via software. This interface has 20-bit converters and pads the LSB to fill a 24-bit file. Conversely, the output of 24-bit sessions is truncated to 20 bits: the *Dither* plug-in allows you to convert a 24-bit signal to 20 bits more elegantly.

1622 I/O. Once you get used to the power of TDM mixing, it's natural to want to insert the audio from your MIDI gear and outboard effects into your Pro Tools system. The 1622 I/O ($1,595) is designed specifically to accommodate this.

The 1622 I/O offers 16 inputs and 2 outputs, all on balanced ¼-inch TRS jacks. Two of the inputs are on the front panel for easy access. The 1622 I/O has the same 20-bit converters as the 882/20 I/O does. It is calibrated for 14 dB of headroom and includes S/PDIF and Superclock I/O.

However, the 1622 I/O has several features not found on its siblings. In addition to a 3-LED meter on the two outputs, each input channel has a 3-LED input meter. (You get software input metering, as well.) Each input has gain-staging accessible via software, not only –10 dBu and +4 dBV, but also higher, in increments of 2 dB. This allows the inputs to accommodate synths, samplers, and effects without patching through a mixer to get more gain. The unit also remembers all gain settings, allowing it to function as a stand-alone 16-by-2 mixer when your computer is off.

ADAT Bridge I/O. The ADAT Bridge I/O ($1,245) provides 16 channels of Lightpipe I/O for devices bearing the ADAT Optical logo. This interface (see **Fig. 3**) eliminates the need for an ADAT card in digital mixers such as the Yamaha 02R and 03D.

Two channels each of S/PDIF and AES/EBU I/O are provided, and the unit can also perform format conversion from Lightpipe to AES/EBU or S/PDIF, and from AES/EBU to Lightpipe. A stereo pair of analog outputs are included for monitoring.

FLEXIBILITY IS KING

I successfully ran an 888/24 I/O and an old 882 I/O together from the same Mix Core card via an optional Y-cable, and a 1622 I/O from the Mix Farm card. Basically, I used the 888/24 I/O for tracking, stereo out, and effects sends and returns, and the other two interfaces for synth inputs. The connectivity of these interfaces with TDM essentially turns the array into an analog and digital audio patch bay.

Pro Tools includes software that helps you correctly set your interface's levels for your system. Be forewarned that the interfaces are extremely hot to the touch while running.

NEW SOFTWARE TOOLS

Pro Tools 5.0 is a major update to the Pro Tools system and includes a host

of new features. One of the most important additions is the integrated MIDI sequencer. Prior to version 5.0, your options for using MIDI with Pro Tools were either importing MIDI files or synchronizing with a third-party sequencer application.

The basic, no-frills sequencer in *Pro Tools* 5.0 lacks many of the refinements found in other digital audio sequencers. Common features missing from *Pro Tools* 5.0 include an event list, song mode, the ability to record multiple channels of MIDI simultaneously, step recording, and notation. If you are into loop-based music, note that *Pro Tools* doesn't offer a convenient looping tool or have groove quantization, which you'll find in programs like Sonic Foundry's *Acid*. Digidesign plans to add features to the sequencer in future revisions.

The Mac version of *Pro Tools* requires the beleaguered OMS but does not allow you to create multidevice instrument definitions. Unlike some third-party sequencers, however, the preset selection window remains open so you can audition a series of patches before you exit the dialog box. Digidesign created its own MIDI management system on the PC side.

Digidesign's late entry into the sequencer game has an advantage. While digital audio sequencers tend to clock audio to MIDI, Pro Tools clocks MIDI to audio at an unprecedented resolution of 960,000 ticks per quarter note. In practice you're recording and doing basic editing at 960 ppqn. However, you can simultaneously display and edit MIDI and audio data with sample accuracy at any visual resolution in the same window.

Although the high clock rate of *Pro Tools* requires it to be the master, you can slave it to a third-party sequencer via MIDI Machine Control.

The feature set in the TDM version of *Pro Tools* 5.0 and the host-based version of *Pro Tools LE* are very similar. Both have several nice enhancements designed to reduce the amount of clicking you have to do. The Smart Tool intelligently combines the Selector, Grabber, and Trimmer tools depending upon where you put the cursor in the waveform or MIDI block. Also, memory locations can now include zoom settings and other preferences.

The playback view can remain stationary while the tracks scroll, minimizing that pesky page jump found in so many timeline-based programs. However, scrolling on my machine was a bit jerky, even when I set the display to 256 colors.

Because the recent Digi 001 review covered *Pro Tools LE* (see the April 2000 issue of **EM**), from here onward I'll focus on the features that are specific to the TDM version of the program.

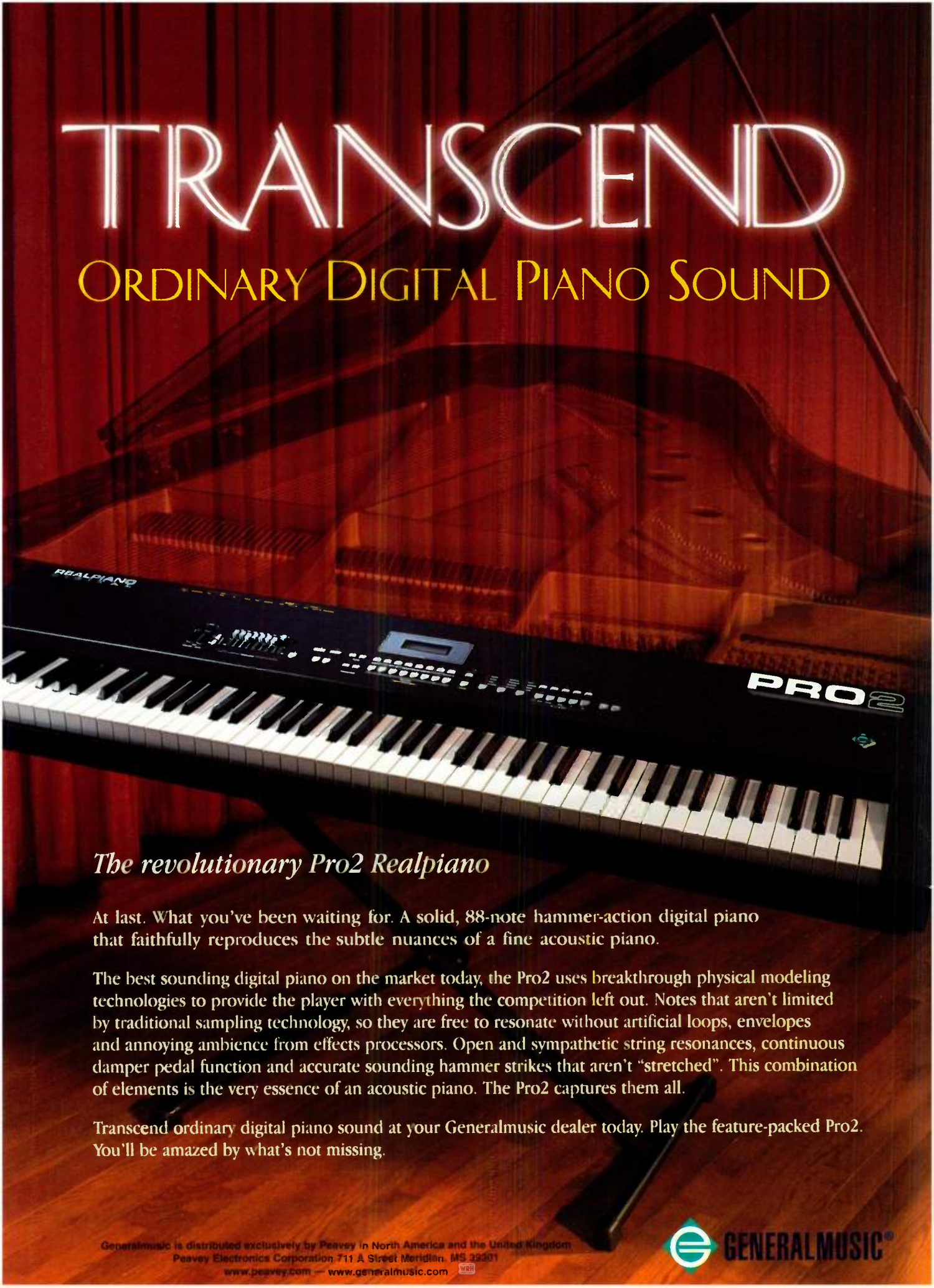
POSTMODERN

The most obvious advantages of *Pro Tools* 5.0 for TDM are card-based DSP, a simultaneous track count of 64 (rather than 24 in *LE*), and 32 buses (instead of 16). Other TDM-only features optimize the system for post-production work. For example, the SMPTE timeline allows you to spot and place events at specific time-code locations. There's also a provision for displaying multiple time rulers and counters, and you can now use the numeric keypad for advanced edit and transport functions.

The Replace Region command lets you replace all instances of one region in the playlist with another region. This particular command is handy for replacing music loops or post-production sound effects.

You can set the Trimmer to automatically invoke the *Time Compression/Expansion* AudioSuite plug-in to scale the length of one region to another region or a video clip. Similarly,

The powerful *SoundBlender*, by Wave Mechanics, lets you combine a variety of effects—including delays and filters—using a modulation matrix.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you can drag a region from the playlist into an edit selection and force it to scale automatically to the selection. (These time-scaling features will also let you match audio segments exhibiting different tempos.) The TDM version outputs MTC and supports MMC.

While Pro Tools hardware and session files are compatible between the Mac and PC, and the software is 99 percent identical between platforms, a few operational differences are worth noting. There is currently no integrated video support on the PC side, whereas Mac users can view QuickTime movies within *Pro Tools* or Avid-compatible video using AVoption hardware (Mac; $7,995).

In *Pro Tools* 5.0.1, the ability to export MP3 files under the Windows NT version has been added (after a 30-day demo period expires, a licensing fee of $19.95 is required for the download). In addition, the NT version has the ability to tag a sound file so that it remains associated with a session, even if you move the files between directories or drives.

DIGIDESIGN
Pro Tools/24 Mix (Mac/Win NT)
digital audio workstation
$7,995
Pro Tools/24 Mixplus
$9,995

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

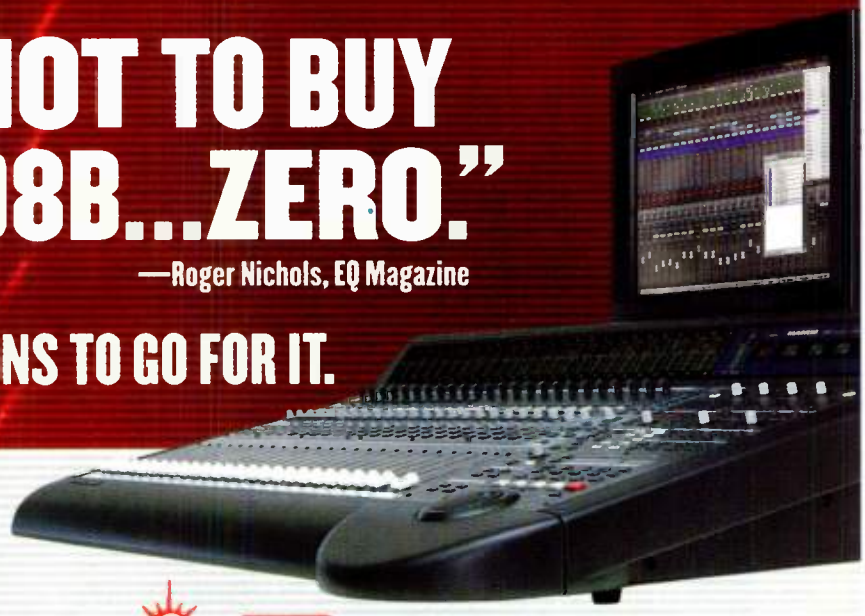
PROS: Offers DSP card–based processing, excellent power and flexibility for routing and mixing, exceptional 24-bit sound quality, 64 simultaneous tracks, 128 simultaneous channels, high clock resolution, cross-platform hardware and session compatibility, and session compatibility with Digi 001 and *Pro Tools LE*. Includes a basic set of plug-ins.

CONS: MIDI sequencer lacks many staples. Provides no tutorial or help within application. Too expensive for many musicians. Interface's Windows support is NT only. The only certified PC system is expensive.

"REASONS NOT TO BUY A MACKIE D8B...ZERO."

—Roger Nichols, EQ Magazine

PLUS 3 MORE REASONS TO GO FOR IT.



1 FREE UPGRADE! NEW OS 3.0 ADDS OVER 30 NEW FEATURES!

Our Programming Department has been chugging the double lattés to create Mackie Realtime OS™ Version 3.0, packed with more new features and enhancements than you can shake a mouse at. Here's just part of what 3.0 adds to the already amazing D8B.

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- Externally or internally accessible **inserts across Mains and Buses** plus **channel inserts pre and post DSP**.
- **Updated GUI** including 48-channel fader bank view screen.
- **Time Offset** (delay) adds a delay of up to 999 samples to the signal at the pre-DSP (dynamics / EQ) point in the signal path.
- **New surround capabilities** including depth-of-center control (LCR mixing with divergence), multiple surround panner window, individual LFE channel level control.
- **Multiple direct outs** per channel.
- **Optional level to tape** fader control.
- **Assignable, bidirectional MIDI control** of all parameters.
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2 DSP PLUG-INS!

Antares' Auto-Tune for the D8B uses advanced DSP algorithms to detect the incoming pitch of a voice or solo instrument as it's being tracked and instantly pitch-correct it without introducing distortion or artifacts. Fully automatable.

Massenburg Parametric EQ. MDW 2x2 High-Resolution Parametric Equalizer plug-in from Grammy-winning engineer/producer George Massenburg. Mono/stereo EQ at 96kHz sample rate for unprecedented clarity and high frequency smoothness.

Drawmer offers two dynamics packages for the D8B: ADX100 includes their industry standard frequency conscious gating, plus compression and limiting; ADX200 adds variable "Peak Punch" and further Drawmer innovations.

IVL Technologies' VocalStudio provides real time vocal doubling, multi-part harmonies and pitch correction in an easy-to-use interface. A free demo is built-into the Digital 8 • Bus. Just add a second MFX card to own this innovative plug-in from a world leader in vocal processing.

TC Electronic Reverb (bundled with the D8B UFX card) provides Reverb 1 and Reverb 2 algorithms from the renowned TC Electronic M2000 Studio Effects Processor. TC FX upgrade package contains an expanded set of M2000 reverbs plus Delay, Chorus, and Pitch. TC 2000 adds the TC M2000's Reverb 3, de-essing, tremolo, phasing, and panning.

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3 1999 TEC AWARD WINNER!

Normally we don't name competitors in our ads. But in this case, Mix Magazine published the other nominees for the 1999 TEC Award for Outstanding Technical Achievement in Small Format Consoles: Allen & Heath's GS-3000, Digidesign's ProControl, Panasonic's WR-DA7, Spirit's Digital 328 and Yamaha's 01V. Thanks to all who helped us win this prestigious award.

SEE US AT NAMM BOOTH #3475

K O R G

D16

Korg's second-generation portable digital studio offers a mighty wallop in a tiny package.

By Bob O'Donnell

If anyone had asked me just a few years ago whether I thought I would ever be able to bring a complete digital recording studio into bed with me, I would have thought they were more than a little weird. Amazingly enough, however, I found myself doing just that while reviewing Korg's D16 Digital Recording Studio. (My wife thought that I was pretty strange for doing so, but that's a story for another day.)

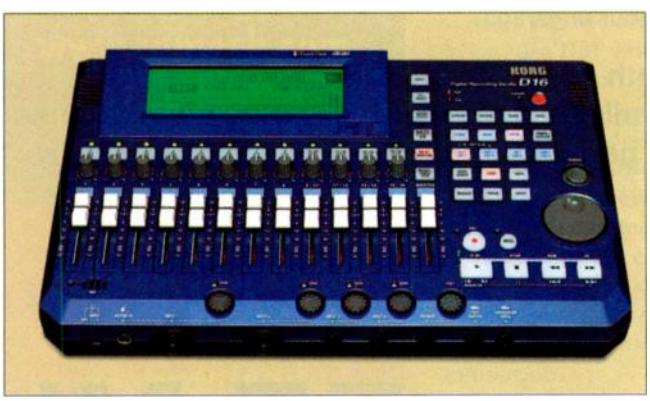
The cobalt blue D16 is impressively compact and lightweight, managing to pack in almost everything you need to record professional-quality, 16-track digital recordings anywhere that you can find power. Topping the list of features is a graphical touchscreen interface to control the device, complete with waveform editing. You'll also find a multiplicity of built-in effects (up to 11 are available at once), a good assortment of analog and digital inputs, a 24-channel mixer, an internal 2.1 GB hard drive, and an external SCSI connector. Heck, it even includes drum patterns and has a built-in microphone!

Korg clearly learned a great deal in producing the D8, its first portable digital studio. With the D16, the company has addressed the D8's shortcomings, taken all of its good ideas, and added more features. The end result is one impressive recorder.

FIT AND FINISH

Pulling the D16 out of the box, the first thing you notice about the unit is how small it is. Weighing only 4.4 pounds and measuring just 14 inches wide by 2 inches high by 9 inches deep, the D16 seems almost too small to contain its own feature set. However, as a result of its tiny size, you can bring it to bed for some late-night manual reading or tote it to a gig or practice session.

Other than its size, the most salient element of the device is its large, angled TouchView LCD. Based on an updated version of the touchscreen technology found in some of Korg's high-end synthesizers, the D16's backlit 240-by-64-pixel display provides a wealth of useful feedback. (And yes, it does have a dedicated contrast knob on the back.) On a piece of equipment as full-featured and complex as the D16, the type of screen and the information that it provides go a long way toward determining the recorder's overall usability. Thankfully, in this regard the D16 delivers in spades. The TouchView screen is nicely designed and very easy to read. In fact, the only personal digital recorder screen that surpasses the D16's is the larger display found on Roland's more expensive VS-1680 (although it's not a touch screen).



The Korg D16 Digital Recording Studio is a massive upgrade of its younger sibling; it adds 24-bit recording, twice as many tracks, and more, while retaining the slick touchscreen interface and the same pint-size dimensions as the D8.

In addition to the large screen, the D16's interface is driven by numerous dedicated function buttons that are located above the unit's transport controls. You can choose from a wide range of system settings and parameters simply by selecting one of the rubbery buttons, and then touching the screen. If you prefer, you can use the unit's cursor control as well. You can also flip through multiple pages within a particular area by simply pressing the dedicated function button repeatedly. All in all, it's a fast and intuitive interface that will help keep the equipment out of the way when all you want to do is record. Still, this is a deep, feature-laden device, so plan to keep the manual by your side until you fully grasp the machine's way of working. Sadly, the included documentation is only mildly helpful in figuring out the D16—a tutorial video would be a great aid.

The D16's front panel features a master fader, transport controls, and the now ubiquitous jog wheel for making editing adjustments. Rounding the panel out are 12 input faders; the first 8 are mono, and 9 through 16 are stereo pairs. Faders 1 through 8 each have a pan control, and the stereo pairs each have a balance knob.

GET CONNECTED

When it comes to connecting the D16 to the rest of your gear, you'll find the device to be very well equipped. Across the front are four balanced analog inputs with dedicated trim controls: two are traditional ¼-inch jacks, and two feature the combination XLR/¼-inch plug design that lets you use either type of connector. In addition, the first input also offers a dedicated ¼-inch guitar jack that's designed specifically for direct recording of electric or acoustic-electric guitars.

Unfortunately, neither of the two XLR connectors offers phantom power, which would be convenient for using the low-cost condenser microphones that are now available for home or personal studio use. Other connectors on the front of the device are a ¼-inch headphone jack, a footswitch jack, and a ¼-inch expression pedal jack that can be used to control the device's built-in effects in real time.

The back panel adds four more balanced analog ¼-inch input jacks with dedicated trim controls, optical S/PDIF stereo inputs and outputs, a ¼-inch mono aux send, and stereo RCA master outs. Another pair of stereo RCA monitor outs has its own dedicated level control. MIDI devices are connected with MIDI in/out jacks. A 25-pin SCSI connector can access external hard disks or removable media, as well as

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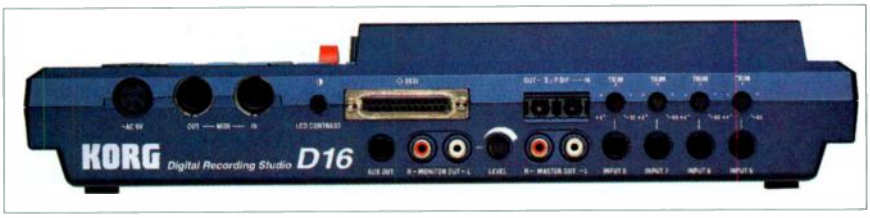

FIG. 1: The rear panel offers a pair of monitor outputs with a dedicated level control. The SCSI chain allows the connection of up to seven devices, including CD-R and CD-RW drives.

CD-R/CD-RW drives (see **Fig. 1**). Connected SCSI devices can be used for additional recording or for backup. However, note that backing up to CD-R is not supported in the first version of the D16's operating system, though Korg originally promised that it would be. Fortunately, Korg plans to release a free software upgrade this summer that will address this problem and add many useful features (more about the upgrade later).

All told, the D16's connection options are broad enough that most people can plug in all the components they've got without needing to switch anything when moving from recording to mixdown. I'm particularly fond of the monitor outputs and dedicated level control, as simple as they may be. On similar units, finding out how to set up and control your monitors can be inordinately difficult. I would have liked a few more XLR mic inputs in case I wanted to record, say, two singers and a guitar at once; but otherwise there's plenty of room to grow.

Another "connection" that deserves mention is the D16's built-in microphone, which offers a three-way switch that lets you turn it off or set it to input 1 or 2. One of the D16's niftiest features is a built-in chromatic tuner that works with plugged-in devices and purely acoustic sources. In order to make it work with acoustic instruments, however, Korg had to build in a microphone. Once it did, the company quickly realized that the same simple dynamic microphone could be used for recording quick audio notes or reminders (sometimes called *slating*) and for recording sudden inspirations, all without the need to plug anything in. The built-in mic certainly won't replace any of your dedicated mics, although you can run it through the D16's microphone modeling effects for added punch (more on this later). Nonetheless, many people will find the built-in mic to be a very handy addition.

ENLIGHTENED DISK COURSE

The recording process on the D16 is straightforward and is similar to that of other portable digital studios. You assign an input to a mixer channel via a dedicated Input page, and that mixer channel is hard-wired to the recording track of the same number. With the D16, you can record up to 16 tracks of uncompressed 16-bit, 44.1 kHz digital audio (8 tracks can be recorded simultaneously). With uncompressed 24-bit, 44.1 kHz recording, the number of physical tracks drops down to 8, with 4 tracks that can be recorded at once. Because the D16's A/D and D/A converters operate with 24-bit precision (the internal mixer's processing is 32-bit), this is a true 24-bit recording mode, unlike the pseudo 24-bit setting found on Roland's VS-1680.

Each input can be routed through the mixer's 3-band sweepable mid EQ and through one of the dedicated insert effects before it's recorded to disk. If you want to work with an external signal processor, you can use the mixer's Aux Send control. The EQ controls offer basic sound shaping and allow you to invert the phase of an input signal, but I found their onscreen layout to be a bit confusing. I would have really liked one additional assignable parameter knob per mixer channel that could be used to quickly tweak EQ or other settings.

Once you have all the input levels, EQ, and effects set up, you can store your mixer settings as a Scene. Up to 100 Scenes are available per song, and you can manually or automatically switch between Scenes during playback, recording, and mixing. For example, you can automate Scenes from a sequencer via Program Change messages, but the D16 doesn't support any kind of dynamic automation—not even via external MIDI continuous controllers—which is unfortunate.

However, Korg plans to release a free upgrade to version 2 of the OS this summer. The new software will add the ability to send and receive all mixer parameters, including panning and fader controls, via MIDI Continuous Controller messages. Another useful track-editing feature in the upgrade will be a Fade In/Fade Out command. The software will also allow the D16 to import and export track data as WAV files. The upgrade will be available on CD and as a download at the Korg Web site.

To simplify the recording process, the D16 has many of the features that you would expect in a well-equipped personal digital recorder, including autopunch in and out, loop recording, and trigger recording. In addition, the D16 offers up to 8 virtual tracks per

D16 Features

Tracks (simultaneous recording)	(4) 24-bit; (8) 16-bit
Tracks (simultaneous playback)	(8) 24-bit; (16) 16-bit
Tracks (virtual)	128
Faders	(8) mono tracks; (4) stereo pairs; (1) master L/R
Songs	100 per drive
Locate Points	4 per song
Mark Points	100 per song; nameable
Levels of Undo	99
Analog Inputs	(2) balanced ¼"/ XLR (combo); (6) unbalanced ¼"; (1) unbalanced ¼" guitar input
Analog Outputs	(2) RCA master; (2) RCA monitor; ,(1) ¼" aux send; (1) ¼" TRS headphone
Digital I/O	stereo S/PDIF optical
Other Ports	MIDI In/Out; (1) ¼" footswitch; (1) ¼" stereo phone jack (expression pedal)
Special Features	(1) built-in dynamic microphone

For manufacturer information, please see Contact Sheet, p. 205.

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physical track. Thus, you could have up to 128 virtual tracks per song, although you can only play back 16 physical tracks at once. Nonetheless, for recording multiple takes of solos and other parts, virtual tracks are a godsend. The ability to easily view (with adjustable peak hold times) prefader and postfader input and track levels is another of the D16's recording amenities, as is the separate cue mix for monitoring. The unit also has a dedicated Solo button and a flexible architecture that lets you quickly hear a single track or groups of tracks during the recording, editing, and mixing process.

ART AND ARCHITECTURE

Two of the D16's most exciting features are its extensive array of built-in effects and its overall effects architecture. The device features 192 preset effects organized into different categories, and includes room for 192 additional user effects. You can also apply up to eight independent insert effects, two independent master effects, and one dedicated final effect for mixdown. As a result, up to four different multi-effects programs can be applied to an audio signal. Each of these programs can consist of several individual effects—the insert effect as well as one or both of the two master effects (depending on how you adjust the appropriate mixer channel's Effects Send 1 and 2 settings). Lastly, the final effect can be applied to the overall mixer output. This scenario may seem like overkill for many recording applications, but it shows that the D16's effects architecture is flexible enough to do just about anything you want it to. You can apply any of the effects to an input channel or a playback track, so you have the option to record a track with signal processing or to record dry and then experiment with different effects later.

As impressive as this arrangement may be, it can be confusing, and the effects architecture does have some limitations. Specifically, the Insert Effects are available in four different modes: one mode offers eight mono inputs and eight mono outputs, another uses four mono inputs and four mono outputs, a third has two stereo inputs and two stereo outputs, and a fourth mode yields two mono inputs and two stereo outputs. Like any other digital device, the D16 has a fixed amount of processing bandwidth. Consequently, the variety, quality, and depth of the effects programs vary according to the different modes. In the 8-in/8-out mode, for example, the 20 available effects programs are limited to single effects, whereas the 40 programs available in the 4-in/4-out mode typically feature two effects per program. In the two different dual-input modes, the 68 available programs are all complex multi-effects, each consisting of several independently adjustable individual effects.

▼

The D16 seems almost too small to hold its own features.

The bottom line here is that if you want to use the more complex multi-effects chains, you'll be limited in the number of insert effects that you can use at once. In practical use I don't believe this to be a serious limitation—you can always record a few tracks dry and then apply the complex programs as insert effects later in the mixdown process.

The available effects sound first-rate and include all the types of effects that you'd expect in a modern digital recording device: reverbs, delays, chorus, flanger, distortion, dynamics processing, and pitch-shifting. There are some other effects that you might not have expected, such as the microphone and guitar-amp simulation models that Korg refers to as REMS (Resonant structure and Electronic circuit Modeling System). These models attempt audio alchemy by taking sounds from inexpensive microphones or direct-injected guitars and converting them into signals that emulate characteristics of more expensive microphones or popular guitar amplifiers. I'm not sure that the emulations are exact duplicates, but there's no doubt that these modeling effects are useful and offer good-sounding, intriguing ways to treat plain vocal and guitar tracks.

In addition to the modeling effects, the D16 offers several useful options, such as multiband dynamics compression and other mastering-type effects. Each of the effects offers a wide range of editable parameters, and many have the ability to control certain parameters in real time, either with the expression pedal jack or with MIDI continuous controllers—a nice touch.

GROOVE 'N' SYNC

The D16 carries over some of the neat rhythm and synchronization ideas originally introduced in the D8. First, it offers 215 built-in preset drum-machine patterns. These patterns can't be edited, but they cover a wide variety of musical genres. They're tremendously helpful, either for use as a metronome with a groove or for songwriting; the audio quality of the drum samples in the D16 is good enough to record. It would be great if you could create and edit your own patterns, but presets are better than nothing.

Another cool feature carried over from the D8 is support for tap tempo. For example, you may want to record

D16 Specifications

A/D Converters	24-bit, 64× oversampling
D/A Converters	24-bit, 128× oversampling
Internal Processing	32-bit (mixer section)
Sampling Rate	44.1 kHz only
Frequency Response	10 Hz–20kHz, ±1 dB (@+4 dBU)
Signal-to-Noise Ratio	100 dB (nominal)
Dynamic Range	100 dB (nominal)
Total Harmonic Distortion	0.02%
Maximum Storage Capacity	2.1 GB (internal); 1,000 GB per drive (external, up to 7 drives)
Display	240 × 64–pixel backlit LCD with touch panel
Dimensions	14" (W) × 2" (H) × 9" (D)
Weight	4.4 lbs.

NemeSys GigaStudio 160

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Soundscape Mixtreme PCI

Based on a completely different concept from most audio soundcards, the Mixtreme™ provides 16 channels of 24 bit digital I/O via 8 channel TDIF ports and also provides an expandable digital mixing environment. Many cards have DSP chips on-board, but the Mixtreme™ uses it's DSP as the engine for effects plug-ins. You get the same quality effects found in the 5 figure systems, without taxing your computer's CPU. The Mixtreme™ provides multi-client and ultra-low latency performance with the most widely used sequencers, audio editors and software synths on the market. Also available form Soundscape are hardware/software bundles, AD/DA converters, and pro studio systems.

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

PROS: Offers uncompressed 16- or 24-bit recording, easy-to-use touchscreen interface, extensive built-in effects, built-in drum patterns, and handy mixdown options.
CONS: Provides no phantom power. Advanced synchronization options have limitations. Lacks dynamic automation.

an acoustic guitar track and then sweeten your tracks with MIDI parts. You can tap along with the beat of the acoustic guitar track and automatically generate a MIDI tempo map that can drive your sequencer and keep it in perfect sync. If you prefer, you can also record a tempo map from a sequencer into the D16 and then use it to drive the D16's metronome, rhythm machine, and the optional bar:beat display.

One minor but notable limitation is that in the original version of the OS, the D16 can send MIDI Time Code (MTC) but does not receive it—the D16 must serve as the master clock. Conversely, although the D16 can respond to MIDI Machine Control (MMC) messages, it does not generate them. Again, the release of Korg's version 2 software will enable the D16 to send and receive both MTC and MMC messages.

EDITING POWER

Like other digital recorders, the D16 offers a wealth of editing options, including the standard ability to copy, insert, erase and delete portions of any track or virtual track. It can also swap, reverse, and optimize sections of audio. The D16's time-compression and expansion options are useful for slightly shortening or lengthening a single track or several tracks at once, as well as for adapting drum loops or other sampled phrases for use in a song.

The D16 includes a dedicated scrub button that you can use to fine-tune your edit points. When you select it, the screen switches to a waveform display of the active track. In addition, many of the edit functions include a Wave button that lets you view and set edit locations in finer detail. You can vertically or horizontally zoom into the waveform display, although you won't have the level of detail or control that you can get from computer-based audio-editing software.

MIXDOWN MAGIC

When it comes time to create a finished stereo product, the D16 offers several useful options. First, you can use one of two special bounce modes that let you record the final stereo mix on the D16 itself—either by mixing 14 tracks down to 2, or by mixing 16 tracks onto 2 virtual tracks. In addition, you can mix up to 24 independent signals at once by using the D16's Sub Input feature. This enables you to add up to 8 live or sequenced tracks to the 16 onboard tracks on mixdown. Many home and personal studio recordists have been dreaming of this kind of flexibility—it's nice to see Korg deliver it.

SIZING IT ALL UP

Despite strong competition in the portable digital studio market, the D16 can hold its own against virtually any of its competitors. It doesn't offer all of the professional niceties that you'll find in the VS-1680, such as phantom power, a larger display, and advanced synchronization options. However, it has many features that its competitors can't touch, including true 24-bit recording options, built-in drum tracks, and an extensive and flexible effects architecture.

The audio quality of the D16 is superb, and its large touchscreen makes it easy to use. I have a few minor feature quibbles with the device, but they can all be fixed in software, and Korg appears to be already working on most of them. If you're looking for a home recording device that you can live, learn, and grow with, the D16 is an excellent choice.

Former **EM** *editor* **Bob O'Donnell** *is a computer-industry analyst at IDC, the author of* Personal Computer Secrets *(IDG Books), and the host of the* O'Donnell on Computers *radio program. His Web site is www.everythingcomputers.com.*

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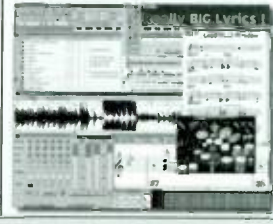

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EGO SYSTEMS

WAMI RACK 24 (WIN)

High-res audio, lots of MIDI, and sync to boot.

By Brian Smithers

Just when you thought you had a handle on the digital audio hardware scene, along comes Ego Systems (Ego Sys), a Korean company with a new line of hard disk recording products aimed squarely at the hearts and budgets of personal-studio owners.

The WaMi Rack 24 takes its name from its split personality—it offers both Wave and MIDI functions. Given its digital I/O and sync capabilities, it could have been called the WaMiS/PdiSmpt Rack, but fortunately the marketing folks prevailed. The WaMi Rack combines a compact PCI card with a 1U rack-mountable interface box that is connected by a 44-pin cable (which the documentation carelessly refers to as a "parallel" cable).

Combining four channels of analog input with eight channels of analog output, the WaMi supports 24-bit audio at up to 96 kHz (as of its latest driver release). It also offers stereo digital I/O, 64 channels of MIDI I/O, and synchronization capabilities. Up to four WaMi Racks can be linked together for synchronized operation if you have enough PCI slots (and an available IRQ for each). A Mac version is currently under development.

BACK OF THE RACK

The WaMi's connections fill the entire back panel of its 9-inch deep rack unit; the sensible layout is compact without being cluttered (see **Fig. 1**). All analog ins and outs use ¼-inch TRS connections, allowing +4 dBu balanced or –10 dBV unbalanced operation. Putting at least one pair of analog inputs on the front panel would have made connecting and disconnecting mic cables more convenient and would also have provided enough space for XLR connectors. However, cables hanging down the front of your rack would make for a less tidy arrangement, so many users will consider this configuration a reasonable trade-off.

The four analog inputs all feature mic- or line-level operation and +12V phantom power. Both the phantom power and mic preamps are selected for pairs of inputs within the WaMi's software control panel. The eight analog outputs are globally selected for +4 dBu or –10 dBV output level. Stereo S/PDIF I/O is available in both coaxial and optical formats. When selected, digital input replaces analog inputs 3 and 4. Analog outputs 1 and 2 are considered the main analog outputs and are therefore always mirrored by the digital ones.

The WaMi reads and generates SMPTE time code through the usual pair of ¼-inch jacks and also slaves to or generates standard word clock and 256× Superclock on BNC connectors. Four MIDI In and four MIDI Out connectors complete the array of jacks on the WaMi's back. Its front is simplicity itself, with status

FIG. 2: *Control Desk* provides access to all of the WaMi's settings, from output level to mic preamps and phantom power. The sample rate can be set manually or automatically, and each pair of inputs can be monitored in real time through any pair of outputs.

FIG. 1: Ego Systems' affordable WaMi Rack 24 is chock-full of I/O, including four analog inputs, eight analog outputs, four MIDI inputs, four MIDI outputs, word clock, SMPTE, and two types of S/PDIF.

For manufacturer information, please see Contact Sheet, p. 205.

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lights for power, MIDI activity, and SMPTE activity, as well as an 8-segment output-level LED for each of the eight outputs.

UP AND RUNNING

Installing the WaMi Rack 24 is like falling off a log, but there's one little thing to watch out for: the drivers don't like to be updated, and Ego Sys recommends removing them before installing the newest version. To this end, the company thoughtfully provides a utility called *infclear.exe*, which can also be found on its Web site as *egoclear.exe*. Fortunately, I noticed this ahead of time and downloaded the latest drivers before installing the hardware. Instead of using the supplied floppy disk, I directed the Windows New Hardware wizard to the downloaded driver. It worked perfectly, and the unit was fully operational in minutes.

▼

The WaMi has a respectable set of synchronization functions.

Ego Sys provides no software bundle with the WaMi Rack 24, and the logic of this decision depends on your perspective. The inclusion of even a lite version of a digital audio editor or sequencer would add great value if you were putting together your first system. On the other hand, if you've already made a significant investment in software, bundled programs are merely a waste of money. Because the WaMi Rack 24 is likely to appeal to musicians from both camps, it's hard to criticize Ego Sys's decision.

Drivers are included for Windows MME, DirectSound, ASIO 2.0, EASI, and NemeSys's GigaSampler. Also included is *Control Desk*, an applet for setting the WaMi's parameters (see **Fig. 2**). The center section of *Control Desk* determines how the analog inputs behave, allowing you to choose among mic level, +4 dBu, and –10 dBV.

WAMI'S LITTLE SIBLING: THE WAMI BOX

I'm a notebook nut, so I was eager to get my hands on Ego Systems' WaMi Box ($600), a PC Card audio and MIDI interface for notebook computers. Notebook musicians have long cried out for quality audio and MIDI for their portable studios, and the WaMi Box delivers the goods. For starters, it's the first notebook device to offer more than two channels of output, with four analog outputs and two inputs. It also provides both coaxial and optical S/PDIF I/O.

On the MIDI side, the WaMi Box has 16 MB of sample RAM compatible with the Soundbank format, and it also includes a 4 MB GS wavetable sound set. The accompanying CD-ROM includes a shareware application called *Virtual Sampler*, and Ego Sys says that it intends to bundle the full version in the near future. *Virtual Sampler* adds real sampler functionality to the WaMi Box, and it also lets you import Creative SoundFonts and files from Akai CD-ROMs.

The PC Card connects to a breakout box about the size of a sandwich and only slightly heavier. The breakout box houses sample RAM, the A/D/A converters, a DSP chip, and all the physical connectors. Analog I/O is on gold-plated RCA jacks with a ¼-inch unbalanced mic input and a ¼-inch stereo headphone output. The only side of the breakout box not filled with connectors sports a row of status lights.

Drivers for ASIO 2.0, EASI, and NemeSys *GigaSampler* are in the works, and there's talk of eventual Macintosh compatibility. Windows 2000 support is also planned for the entire Ego Sys product line starting with the WaMi Box. In addition to DirectX-compatible Windows drivers, the bundled software includes a *Mixer* applet (see **Fig. A**) and a Soundbank manager. The *Mixer* provides faders for MIDI output and audio input and output levels, with solo and mute buttons for each. It also supplies controls for the WaMi Box's DSP-based effects, which include seven reverb types and nine chorus types. The effects are not controllable from your digital audio sequencer, however.

FIG. A: The *Mixer* applet in Ego Systems' WaMi Box controls the input and output levels and also provides hardware-based chorus and reverb. The Box offers two audio inputs, four audio outputs, and 16 channels each of internal and external MIDI.

Despite the fact that it's billed as 20 bit, the Box can only handle 16-bit audio—the 20-bit designation refers to its converters. It's also hardwired at 44.1 kHz, so if you import 48 kHz audio through the S/PDIF input, it will be converted to 44.1 kHz on the fly. The conversion algorithm sounds pretty good, though, so that's a minor flaw in an otherwise excellent product.

I had a lot of fun with the WaMi Box; it's a great asset for anyone trying to get a lot of musical mileage out of a laptop computer.

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WaMi Rack 24
Minimum System Requirements
Pentium/166; 32 MB RAM; Windows 95/98/98 SE; PCI slot

You can select balanced or unbalanced operation and turn phantom power on and off by clicking on the applet's clearly marked buttons. Above the buttons are four input level knobs, which can be ganged to control inputs 1 and 2 or 3 and 4 as a stereo pair.

Along the left edge of *Control Desk* are buttons for setting the clock source and digital I/O. The digital input features hardware sample-rate conversion, which is engaged or defeated by a large button sensibly labeled "Digital Input with SRC." (For transferring bit-for-bit clones of course, you would defeat the sample-rate conversion.) Below that, another group of buttons sets the source for inputs 3 and 4 as either analog, coaxial S/PDIF, or optical S/PDIF. Farther down the left side, you find selections for word clock, clock source, and consumer- or professional-format digital output.

You select the sample rate with a set of buttons in the upper right of *Control Desk*. If you select Auto, WaMi adjusts to the settings of your digital audio software when recording from the analog inputs, or adjusts to the format of the digital input when recording digitally. The Input Monitor section lets you pipe a copy of input pair 1 and 2 or 3 and 4 directly to any pair of outputs for real-time monitoring without the latency introduced by first running the audio through the computer. This is a great feature; but the addition of a level control to the input monitor would be ideal, allowing you to mix the signal with other outgoing audio without affecting input record levels. The best way to accomplish that with the WaMi is to assign the input monitor to its own output pair and then to mix it with the other outputs through an external mixer.

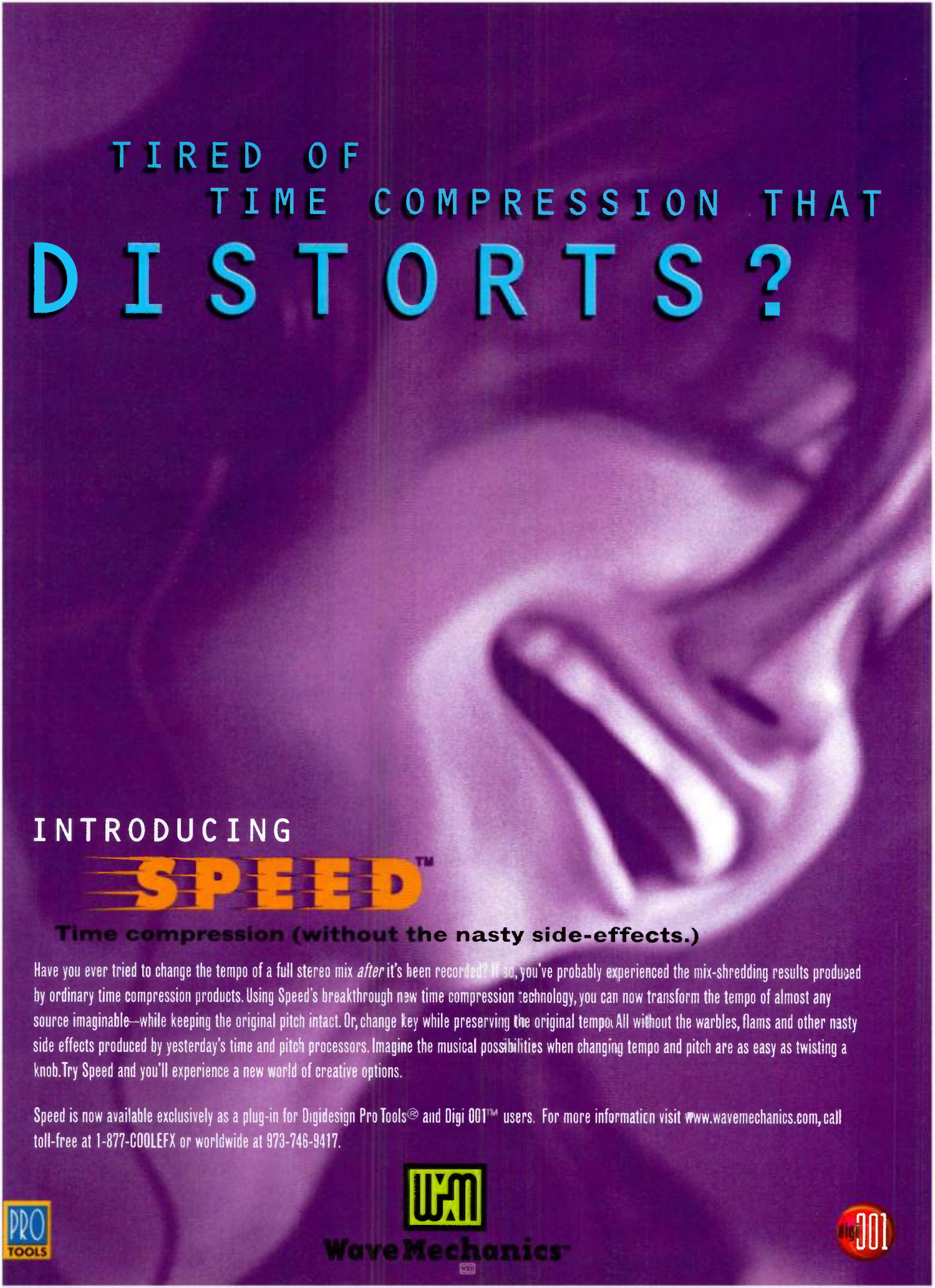
To prevent clipping when multiple signals are sent to the input monitor, *Control Desk* provides two buttons in the Mix Mode section. By selecting Soft, you tell the WaMi to reduce the output monitor levels by 6 to 12 dB depending on how many input signals are present. That allows you to set hot

EGO SYSTEMS
WaMi Rack 24 digital audio and MIDI interface
$600

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

PROS: Easy to use and offers great sound at a reasonable price. Features 24-bit, 96 kHz audio support; mic preamps on all four inputs with phantom power; word clock; SMPTE; and 64 MIDI channels. Up to four units can be linked for 16-In/32-Out operation and 256 MIDI channels.

CONS: Loud popping sounds on power-up and power-down. Phantom power is only +12V. Provides no software bundle or headphone output.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WaMi Rack 24 Specifications

Analog Inputs	(4) balanced ¼" TRS; mic or line level (+4 dBu or –10 dBV), selectable in software; +12V phantom power
Analog Outputs	(8) balanced ¼" TRS; +4 dBu or –10 dBV, selectable in software
Digital I/O (stereo)	(1) S/PDIF coaxial or optical
A/D/A Converters	24-bit, 96 kHz
Sampling Rates	32, 44.1, 48, 64, 88.2, 96 kHz
Synchronization	word clock/Superclock In/Out via BNC connectors; SMPTE time code via ¼" connectors
Frequency Response	10 Hz–20 kHz
Dynamic Range	120 dBA
Signal-to-Noise Ratio	120 dBA
MIDI I/O	(4) MIDI In; (4) MIDI Out
Dimensions	1U × 9" (D)

input levels without overdriving the output levels. If you want to bypass this setting, simply select Dynamic. Just below the Mix Mode section is a pair of buttons that can be used to set the level of the eight analog outputs for +4 dBu balanced or –10 dBV unbalanced operation.

For a device in its price range, the WaMi has a respectable set of synchronization functions. It can slave to another device through its S/PDIF input or through word clock or Superclock. It can even get its clock from the digital input while recording through analog inputs 3 and 4. It also features SMPTE time code I/O. A little pop-up panel at the bottom of *Control Desk* called SMPTE Desk lets you send or receive time code at 24, 25, 29.97, 30 drop, or 30 frames per second. You can stripe a tape easily and even set an offset. WaMi translates incoming time code to MIDI Time Code (MTC) for your digital audio sequencer, and the manual includes several pages of background and instructions for synching with external devices.

WAMI, HOW I LOVE YA

The WaMi Rack 24 was well behaved and a pleasure to use. Although I didn't put it on a bench to confirm its almost-too-good-to-be-true claim of 120 dB dynamic range and S/N ratio, I can certainly vouch for the fact that it is very quiet and sounds great. The more closely I listened, the more I found other weak links in my recording chain.

I would have preferred +48V phantom power, but +12V seems to be common in computer-based devices. Because the WaMi has its own power cord, though, you'd think the computer wouldn't be a limitation. The mic preamps had plenty of gain and a pleasingly neutral sound; unless you've got a persnickety condenser mic that shuns +12V phantom power (as mine does), you're in good shape for plugging your favorite mic directly into the WaMi to record. However, you will want to buy or build a mic cable (or four) that terminates in a ¼-inch male TRS plug. That way, you can leave the cable plugged into the back of the WaMi and avoid having to fuss with XLR-to-¼-inch adapters. Ego Sys thoughtfully provides wiring diagrams in the manual for this purpose.

I would like to see a front-panel headphone jack. The WaMi's mic preamps and phantom power obviate the need for a mixer on the input end, so why not complete the circle? My only other quibble with the WaMi is that it pops quite nastily on power-up and power-down. To its credit, Ego Sys warns of this in the manual, but I expect better from such an otherwise fine piece of gear.

With its analog and digital I/O and 24-bit, 96 kHz support, it's easy to think of the WaMi Rack 24 as primarily an audio interface and to forget its other personality. For many people, however, the WaMi's 64 channels of MIDI and ample synchronization capabilities will end the search for a MIDI interface. With so much good stuff at a fair price, the WaMi really packs a wallop.

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KURZWEIL

PC2X

The venerable PC88 is updated with twice as much of just about everything.

By Scott Wilkinson

FIG. 2: The back panel sports an RCA digital audio output in addition to a pair of balanced analog outputs, one headphone output, three footswitch inputs, two footpedal inputs, breath-controller input, and ribbon-controller input as well as MIDI In, Out, and Out/Thru.

Kurzweil has always been well known for its fine electronic musical instruments. For example, the PC88 (reviewed in the December 1994 issue of **EM**) has enjoyed its status as a top-notch master keyboard controller since its introduction. In fact, **EM** editor Steve Oppenheimer used one to lay down the keyboard and bass parts on an album that I recorded with my wife in 1997.

Of course, technology marches inexorably onward, and the PC88 has now been succeeded by the new PC2, which improves on its excellent predecessor by a factor of two in many respects. For starters, it's available in two flavors: the PC2 has 76 semiweighted keys, whereas the PC2x sports 88 fully weighted keys. Of course, both keyboards are sensitive to Velocity and Channel Aftertouch. I reviewed the PC2x; but because the two models differ only in the number of keys and the keyboard action that they offer, I will use the more general term PC2 to discuss them both.

SIGHT

The PC2 embodies Kurzweil's new industrial design—most buttons and controls are located within blue-gray regions on an otherwise dark-purple front panel (see **Fig. 1**). Of course, a pitch-bend wheel and modulation wheel lie to the left of the keys, and two assignable buttons reside just above the wheels.

To the right of the master-volume fader is the Zone Select and Assignable Controllers region, which includes four zone-select buttons, four sliders, a Solo button, an EQ button, and three more assignable buttons. All of these controls serve different functions depending on what you need to do, and they default to certain tasks as indicated by labels of different colors. For example, three of the sliders boost or cut the low, mid, and high bands in EQ mode, and they are labeled Low, Mid, and High in white. In addition, all four act as drawbar controls in KB3 Organ mode (which I'll discuss shortly), for which they are labeled in orange.

The display is a 2-line-by-20-character LCD, which seems very skimpy in an instrument at this level. Two cursor-navigation buttons are located just below the display. Buttons that provide direct access to zone, sound, and system parameters, as well as to the Compare, Copy, and Store functions, lie to the right of the display.

Programs can be selected from one of four banks: Internal, User, and two user-installable expansion banks, all four of which are accessible by dedicated buttons. Each bank holds 128 Programs, which are organized by instrument type into 16 groups of eight sounds each; these groups can also be selected with dedicated buttons. One nice feature in this regard is the ability to set the instrument to return to any given Program in a group when you press that group's button—very convenient. The Sound Select buttons are joined by some System Setup buttons that provide access to the internal voices, MIDI setups, and KB3 mode.

To the right of the Sound/Setup Select region is the Effects and Reverb section, which provides access to two multi-effects processors. At the far right of the front panel is the Data Entry section, which includes increment/decrement buttons, an alpha wheel, and a 10-key alphanumeric pad with Clear, Enter, and Cancel buttons.

The back panel (see **Fig. 2**) includes the power switch and a receptacle for the power cord, which is of the lump-in-the-line variety (presumably to keep the power supply out of the instrument itself). The three MIDI ports are In, Out, and a selectable Out/Thru. There are two balanced analog audio outputs on ¼-inch TRS jacks and a digital audio output on an RCA jack. This output can be set to send AES/EBU or S/PDIF data at 48 kHz with a selectable resolution of 16, 18, 20, or 24 bits. Controller inputs can accommodate three footswitches, two continuous footpedals, a breath controller, and a Kurzweil ribbon controller. A ¼-inch stereo headphone jack rounds out the back panel.

SOUND

The PC2 provides a generous 64 voices of polyphony—twice as many as a stock PC88. (The PC88MX with the VGM General MIDI option provides 64 voices.) In addition, the PC2's polyphony can be expanded to 128 voices with the XP Expanded Polyphony Option (price TBA).

A stock PC2 includes 16 MB of waveform ROM; two optional, user-installable ROM expansion boards (price TBA) add another 16 MB each. All ROM blocks (internal and expansion) also hold 128 Programs apiece, which is twice the capacity of the PC88. A battery-backed user RAM bank holds up to 128 user Programs, a feature which was not available in the PC88.

As mentioned earlier, the Programs in each bank are organized into 16 instrumental

FIG. 1: The Kurzweil PC2 offers twice as much polyphony and twice as many ROM Programs and user RAM Setups as the PC88, all in a new purple case.

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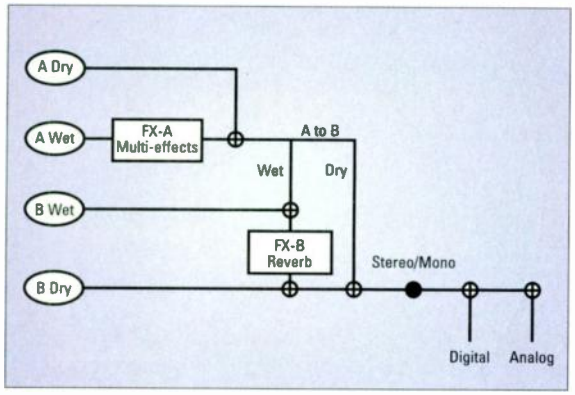

FIG. 3: Each Program sends its signal to FX-A and FX-B, and separate wet/dry controls determine the relative proportions of the signal that are processed. In addition, the PC2 provides separate FX-A and FX-B wet/dry controls for each incoming MIDI channel. Finally, FX-A's output can be routed to the input FX-B in a proportion determined by a global wet/dry control.

groups of eight sounds each. These groups are Piano 1 and 2, E. Piano 1 and 2, Pop Keys, Clavier, Organ, Brass, Strings, Voices, Synths, Pads, Guitar, Bass, Drums, and Percussion. (Actually, these groups apply only to the internal ROM bank; the user bank can have any type of sound in any location, and the ROM expansion banks have different types of sounds.)

Unlike Programs in the PC88, these Programs can be edited and saved in the user bank, although the editing capabilities are relatively rudimentary. Rather than a full-blown synth, Kurzweil wanted to create a preset instrument that you could tweak, and I think the PC2 strikes a fine balance in this regard. Each Program consists of up to four Layers, and each Layer is based on one or two Keymaps (one for mono Programs, two for stereo Programs), which determine the waveform assignment. Of course, multiple, simultaneous Layers reduce the effective polyphony.

Some of the piano Keymaps are limited to certain Velocity ranges, which allows different waveforms to be played in these ranges. These Keymaps are assembled into piano Programs whose sound varies appropriately at different Velocities, which is wonderful. There are even a few single Keymaps with different piano waveforms in different Velocity ranges, which provides a multi-Velocity piano Program without taking up multiple Layers. Either way, this makes for a much more realistic and expressive sound than most sampled pianos can produce.

The Program editor provides access to three groups of parameters for each Layer. In the Timbre section, you can select Keymaps and specify key range, relative volume, several initial controller values, and a few DSP parameters. In fact, each Program can include one or two DSP filters, which can be 2-pole lowpass or highpass with 48 dB resonance; bandpass; notch; shaper; or distortion. Both filters must be of the same type, and some take up both slots with one filter. The Envelope section lets you specify the attack, decay, and release, and the LFO section lets you adjust the rate and waveform of two LFOs as well as the low and high rates of the rotary speaker emulator.

The PC2 includes Kurzweil's KB3 Drawbar Organ mode, which was first introduced in version 2.52 of the K2500 operating system. This mode models the behavior of a tonewheel organ such as the Hammond B-3. The four sliders become drawbars that control the level of sine waves at different frequencies relative to the fundamental. Up to nine sine waves can be used to create an organ sound; the Solo button toggles the sliders between two groups of four, and the mod wheel controls the ninth sine wave.

KB3 mode includes everything needed to accurately emulate a Hammond/Leslie setup, including percussion, key click, chorus, vibrato, tube-amp distortion, and rotary speaker simulation with programmable speed control that ramps up and down in a realistic manner. All of these parameters are accessed for editing by pressing any of the three Sound Parameter buttons while in KB3 mode. There are 16 KB3 Programs in ROM, and another 128 can be stored in user RAM.

The PC2 includes two multi-effects processors—FX-A and FX-B (see **Fig. 3**). These processors are equivalent to

PC2 Specifications

Analog Audio Outputs	(2) balanced ¼" TRS; ¼" TRS headphone jack
Digital Audio Output	RCA (AES/EBU or S/PDIF at 48 kHz and 16-, 18-, 20-, or 24-bit resolution)
Additional Ports	MIDI In, Out, Out/Thru; (3) ¼" TS footswitch jacks; (2) ¼" TRS continuous footpedal jacks; ¼" breath-controller jack; ribbon-controller jack
Polyphony	64 voices (expandable to 128)
Multitimbral Parts	16
Sound Engine	sample playback
Zones	4
Keyboard	PC2: 76-note semiweighted; Velocity- and Pressure-sensitive PC2x: 88-note weighted; Velocity- and Pressure-sensitive
Controllers	pitch-bend wheel; modulation wheel; (4) data sliders; (5) panel switches
Sample-Based Programs (ROM/User RAM)	128/128
KB3 Programs (ROM/User RAM)	16/128
Setups (ROM/User RAM)	32/128
Removable Storage	none
Effects Presets	(163); reverbs, delays, choruses, flangers, phasers, tremolos, panners, filters, distortion effects, rotary speakers, compressors, enhancers, waveform shapers, combinations
Dimensions	PC2: 47.88" (L) × 4.31" (H) × 14" (D) PC2x: 54.31" (L) × 4.31" (H) × 14" (D)
Weight	PC2: 39 lbs.; PC2x: 55 lbs.

For manufacturer information, please see Contact Sheet, p. 205.

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those found in the KDFX for the K2500 and the new K2600. FX-A provides 163 presets based on 56 algorithms, including reverbs, delays, choruses, flangers, phasers, tremolos, panners, filters, distortion effects, rotary speakers, compressors, enhancers, waveform shapers, and many different combinations. FX-B offers 30 reverbs that are also available in FX-A. Each effect provides four editable parameters; that seems a bit skimpy, but it's probably enough for most applications.

TOUCH

The "feel" of a keyboard's action is vital to the success of a master controller, and Kurzweil has been working with Fatar on its keyboard mechanisms for eight years. The PC2x uses Fatar's latest weighted keyboard, which is more rugged than that of the PC88. Obviously, pianists will want to use the weighted PC2x; organists will probably prefer the 76-note semiweighted PC2.

I'm not a piano player, so I invited my friend Nina Goldin over to tickle the weighted "ivories" of the PC2x. Goldin is an excellent piano player who owns a K2500XS, so I knew that she was eminently qualified to evaluate this instrument's action. In addition, we listened to the internal Programs to hear if they "touched" us, as all good sounds should.

Goldin's initial comment was, "It has a nice feel to it." She said that the PC2x felt different than a PC88, with more give than the older model. It was a bit floppier and not quite as smooth or even as the PC88, but it was very easy to play. Goldin also said that the PC2x felt a bit different from her K2500X, which ostensibly has the same mechanism; I chalked this up to individual variations.

The Solo Grand Piano elicited an immediate "Very, very nice!" I liked most of the other acoustic pianos, many of which use multi-Velocity Keymaps. Some of the electric pianos were good in the midrange but a bit thin on top; two notable exceptions were Dyno My E Pno and Big Red Wurly, both of which sounded big and fat throughout

▼

The PC2 improves on its predecessor by a factor of two in many respects.

their entire range. Goldin really liked some of the clavinet sounds, saying they sounded just like her real clav.

I thought that it might be a bit difficult to play organ sounds on a weighted keyboard, but according to Goldin it wasn't a struggle. Many of the organ sounds were excellent, and the KB3 elements (such as the rotating speaker simulator) worked very well. However, Goldin said that the preset organ sounds were a bit too clean overall—not enough grunge. Of course, you can always turn up the distortion if desired. On a different note, I especially liked the Pipe Organ.

Goldin liked the brass and saxophone Programs quite a bit; but being a brass player myself, I found them wanting (as all sampled brass instruments are in my opinion). The strings, on the other hand, sounded absolutely fabulous. "The strings are some of the nicest I've heard," Goldin said. "They're a pleasure to play, and they don't feel synthetic. Kurzweil did an incredible

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SOUND QUALITY ■■■■■
VALUE ■■■■
1 2 3 4 5

PROS: Offers editable Programs, KB3 mode, multi-Velocity pianos, 24-bit digital audio output, and expansion slots.
CONS: Display is small. Provides no disk drive. Some button labels are confusing.

job on the strings; every sound is tasteful and musical." I agree completely.

The vocal Programs are derived from the samples that Take 6 did for Kurzweil, and they also sound great. I like most of the synth sounds very much, and the guitars and basses are similarly excellent. In the drum department, most of the sounds are exceptional, and the percussion sounds are quite good, although the vibes are pretty thin.

The overall sound of the PC2 is crystal clear and clean. Most sounds cut through with no trouble and would make a fine addition to any studio or stage.

CONTROL

Of course, no master keyboard controller would be worth much without extensive MIDI capabilities, and the PC2 delivers. The instrument can be divided into four layered, split, or overlapping zones, and each can send on a separate MIDI channel. A four-zone construct is called a Setup. There are 32 Setups in the internal ROM, and each of the ROM expansions includes 32 more. The user bank can hold up to 128 Setups—twice as many as the PC88.

All controllers can be assigned to different functions in each zone, and all sorts of initial controller values (including Bank Select and Program Change) can be sent when each Setup is selected. Of course, you can assign any Program to any zone, and you can layer or switch between zones using Velocity.

One very nice feature is called Auto-Split; it lets you create a complete Setup with very few button presses. This function offers one user-defined default split point, so you are limited to creating Setups with one or two layered Programs on one side of the split point and one or two layered Programs on the other side. However, this process is much easier than creating a similar Setup in the Setup editor.

The PC2 includes a sophisticated arpeggiator that produces rhythmic melodic patterns when a group of notes is played on the keyboard or enters the instrument via MIDI. There are many parameters, which determine the zones that are affected, the upper and lower limits of notes to be played, the order in which the notes are played, how the arpeggios are latched (turned on and off), the rhythmic values and durations that are used, how notes are shifted up or down in pitch after each cycle, and how Velocity and Aftertouch affect the result. The arpeggiator can sync to its own internal clock or external MIDI Clock messages, and it can also send and respond to System Real Time Start and Stop commands.

Of course, the PC2 is fully capable of SysEx dumps and loads—a good thing, because there is no other offline storage (no floppy drive, hard drive, SCSI port, or memory card). Interestingly, the PC2 is 16-part multitimbral when responding to incoming MIDI data—very cool. Finally, it sends and recognizes all Control Change messages, just as it should.

STOP

For those who want a sophisticated master keyboard controller, the Kurzweil PC2 is hard to beat, especially when you have your choice of a 76-note semi-weighted or 88-note fully weighted keyboard. It's the most expensive instrument of its kind that's currently available, but it offers a wealth of great sounds to tweak, a host of useful MIDI capabilities, and many other features that put it at the front of the pack. And, like many Kurzweil instruments, its operating software is stored in Flash ROM, which means that it can be updated to avoid obsolescence for a long time to come. All you need to do is try one and see if it touches you.

EM *technical editor* **Scott Wilkinson** *is glad to be back on board as a senior member of the editorial team.*

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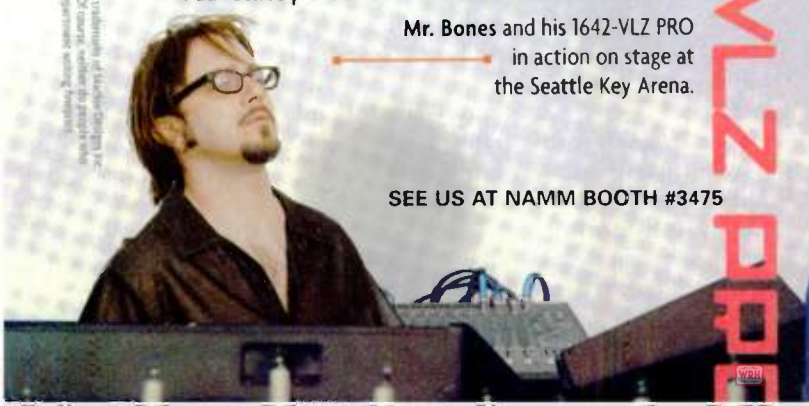

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FXPANSION

SERIES ONE (DIRECTX/VST)

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By Phil Darg

Every once in a while, something comes along and drastically redefines your concept of signal processing. FXpansion's Series One plug-in bundle is such a product, offering fresh effects that color your music in new ways. The plug-ins that make up this groundbreaking and original package have great sonic complexity and can radically shape a signal. Because of their depth of features, using them can be challenging, but the results are definitely worth it.

The bundle contains five plug-ins: *PhatSync, AutopOle, Robotik Vocoder, Ring Modulator,* and *Midi Comb.* (Note: since this review was written, FXpansion has added a sixth plug-in—a multitap delay called *MTap.*) The plug-ins run in Windows VST and DirectX host applications, including the latest versions of Steinberg's *Cubase,* Cakewalk's *Pro Audio,* Emagic's *Logic Audio,* and Sonic Foundry's *Acid Pro* and *Vegas.* They are available separately and bundled, as downloads and on CD-ROM.

PHRESH

The first plug-in I tested was *PhatSync,* a filtering effect for which both frequency cutoff points and resonance are variable. It sounds like a voice from a radio that's being tuned or an electric guitar being shifted through a phaser on extreme settings. Think of it as an extreme EQ with resonance for special effects on instrument and voice tracks. It's also great for fattening up synth-bass notes, enhancing low-frequency definition, and tweaking the midrange.

FIG. 1: Using *PhatSync*'s Pattern Matrix, you can switch from one frequency-and-resonance setting to another either by clicking on a button or sending MIDI messages from your sequencer.

Instead of simply cutting or boosting a frequency, as an equalizer does, *PhatSync* reshapes the frequency content of the entire audio clip according to a user-selected frequency range and adds resonance. The result is a distinctly electronic-sounding prolongation and intensification of the sound that I can only describe as a sort of gliding, modulated electronic spit. The static settings alone are very cool, but you get even more impressive results—such as dynamic frequency shifting and gliding—when you modify the parameters in real time with MIDI.

PhatSync's main controls (a set of dials) are Base Cutoff (frequency cutoff), Base Resonance, Mode Mix, Glide, Mix (wet/dry), and Output Level. You also get two sets of 16-band sliders for fine-tuning the cutoff frequency and resonance. Altering Base Cutoff, Mode Mix, and Base Resonance had the greatest effect, turning the signal from a low and fat rasp into a high and tinny broadcastlike sound.

PhatSync also sports four user memories that store snapshots of the 16-band slider settings; you call up each memory simply by clicking on a button. This feature offers more than quick preset storage—you can arrange your four snapshots in a pattern and activate it with MIDI commands from your sequencer (see Fig. 1).

I tested *PhatSync* on different types of tracks, and in each case I was able to substantially alter the original signal. I was particularly impressed with the way that the plug-in fattened up lower synth notes without causing distortion or overload. It also worked well as a special effect on vocals, and it functioned almost like an amp simulator on electric guitar tracks.

FIG. 2: *AutopOle* uses many of the sound-shaping tools found in synthesizers. The routing-switch buttons in the bottom center of the display control the configuration of the two filters.

THAT '70S SYNTH

AutopOle is more than an effect—this plug-in has many of the controls that synthesizers use to process waveforms (see Fig. 2). *AutopOle*'s envelope generators and LFOs make the audio sound as if it's been run through an old '70s synthesizer.

The main sliders adjust cutoff, resonance, LFO modulation, and EG modulation; and a number of filter buttons affect the overall frequency bias. *AutopOle* uses two envelope generators, each with its own multimode filters. This setup gives you a few signal-routing choices. The routing-switch feature lets you process the left and right channels separately, blend them together, or have one filter carry or process the other. Each configuration offers a distinct sound.

AutopOle can create a wide variety of effects. The results are a lot like what you'd get from using an analog-synth LFO to modulate sounds with variable resonance and frequency settings. Extreme settings completely changed the audio into wah-ing pulses or sci-fi sound effects. The software also generates a useful funky tremolo and wah effect. Apply this plug-in with caution, however—extreme settings can cause harsh clipping.

AutopOle allows you to make the modulation rate dependent on the audio data's amplitude. If you move the EG

For manufacturer information, please see Contact Sheet, p. 205.

NOVA

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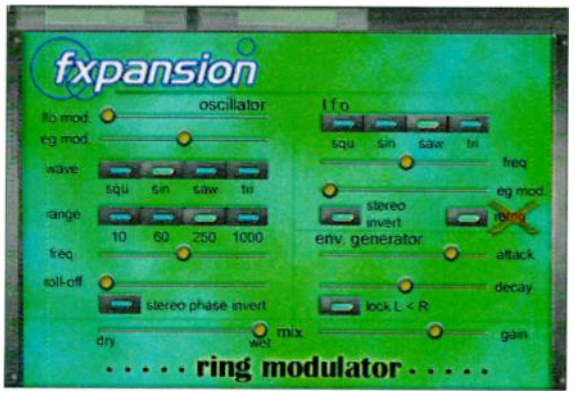

FIG. 3: *Ring Modulator*'s LFO and Envelope Generator can each be fed one of four typical synthesizer waveforms. A virtual LED indicates the selected waveform.

slider to the left, loud signals slow down the oscillation rate; if you move it to the right, loud signals increase the oscillation rate. The resultant signal is more expressive and variable than that produced by a typical chorus or tremolo effect.

MR. ROBOTO

Vocoders have a distinctive sound and are an especially effective treatment for sounds and music with a techno edge. They produce a sound best described as a hybrid of voice and synthesizer, but other combinations of instruments and sounds are possible. FXpansion's *Robotik Vocoder* gets its name from the disembodied, machinelike sound it can give the human voice. Some settings can also create other types of sounds, including ghost voices and deep, phaserlike effects.

Controls include Carrier Gain and Modulator Gain. The carrier and modulator signals require separate inputs. Slew Rate, and Q Factor. The plug-in also offers the LF Widen parameter, which widens or narrows the bandwidth of the low-frequency bands. A high LF Wide setting reduces boominess and softens ringy qualities in the low end. And two sliders control the nature of the noise that the plug-in generates when the modulator track is silent. The sound changed the most when I fiddled with Slew Rate and Q Factor; the former affected the modulation rate, the latter overall frequency response.

Robotik Vocoder's sound quality is good but not pristine. When I compared it with a hardware vocoder, I found that while the plug-in didn't sound quite as good, it handled dynamics better and was easier to control. Of course, a hardware unit costs 20 times as much as the plug-in, so clearly *Robotik Vocoder* is a great value.

I'd like to see the inclusion of some standard waveforms (saw, square, and so on) within the plug-in itself. They're in *Ring Modulator* (which I'll discuss next) but not in *Robotik Vocoder*. The addition would give you a number of carrier-wave choices that you could simply select and apply. As it stands, you must either feed in another audio source (possible only in VST) or use a single stereo track with the carrier and modulator signals on the left and right sides. Adjustable stock waveforms would make the process much easier and quicker; the separate-input option could be reserved for advanced users.

RING-A-DING-DING

Ring Modulator was a real treat and is definitely one of the crown jewels of this bundle. Ring modulators work by using one signal's amplitude to modulate another signal, resulting in a clangorous, metallic sound. They're behind many of the stock sound effects used in film and television. Ring modulators can also create an atonal falling sound, as in Norman Greenbaum's "Spirit in the Sky."

This plug-in has all the controls necessary to create just about any ring modulator sound you can imagine. Its four sections are Oscillator, LFO, Envelope Generator, and Mix (see **Fig. 3**). In the Oscillator and LFO sections you get a frequency slider and a choice of four basic waveforms: square, sine, saw, and triangle. You can also adjust the EG's attack and decay times. A wet/dry mix slider and gain slider complete the array of controls.

I compared *Ring Modulator* with a hardware ring modulator unit and found the plug-in better-sounding, easier to use, and infinitely more fun. I was able to create a number of significantly different effects, from crazy water sounds to dialogue from *Battlestar Galactica*. *Ring Modulator* will keep sci-fi types busy for hours and give musicians a handy special effects tool with endless variations.

MIDI ME

Of all the Series One plug-ins, *Midi Comb* is the most difficult to use. Despite its title, it is not a MIDI-processing plug-in. Although it requires both audio and MIDI data to work, the effect processes only audio data. There are two ways to operate it: either apply existing MIDI data (notes and controllers) from a MIDI track to an audio track, or play MIDI data into the effect as it processes an audio track. The first option is available only in the DirectX version, and to make the process work you need to download and install Hubert Winkler's *Hubi's Loopback Device* or Propellerheads' *MIDI Yoke* (freeware utilities available at www.hitsquad.com).

The *Comb* in *Midi Comb* refers to the comb-filter sound that you create with flangers using deep modulation with short time settings. *Midi Comb* does more than a flanger; it uses MIDI note data to determine the effect's frequency. The result is a MIDI-controlled comb filter in which frequency mirrors the pitch of the note data played on the MIDI keyboard. Other MIDI messages perform different tasks. For example, Control Change 1 (modulation) works like a hard tremolo, opening and shutting the signal completely. In the DirectX version, you can also control *Midi Comb*'s parameter settings in real time with a keyboard

FXPANSION
Series One (DirectX/VST)
effects-processor plug-in bundle
$100

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

PROS: Offers fresh new sounds, gorgeous graphics, and a well-designed user interface. Price is reasonable.
CONS: Doesn't include presets. You can't control VST versions through MIDI. *Robotik Vocoder* and *Midi Comb* require two input sources.

Series One
Minimum System Requirements
Pentium/166 (200 recommended); 32 MB RAM; Windows 95/98/NT 4.0; Windows-compatible sound card

or other hardware MIDI controller.

The main controls are the Amp EG Attack slider, which changes the attack rate, and the Velocity slider, which adjusts the MIDI Velocity control of amplitude. The Highpass Filter and Lowpass Filter sliders control cutoff frequency, and the Feedback slider alters the effect's intensity. Other sliders are for determining the ratio of wet to dry signal, setting the output level, and modifying the effect of Velocity on other parameters.

The resulting sounds are interesting to say the least. *Midi Comb* is a combination of a metallic comb filter, a delay, a short reverb, and variable pitch. A unique feature of *Midi Comb* is its ability to alter the pitch of the affected signal. You can even play multiple notes to create a chord. The effect works well when applied correctly, but it isn't easy to use. To make it work in my DirectX host program, I had to temporarily disable the host software's MIDI input so that the plug-in could receive MIDI data.

CRUNCHY AND GUI

The FXpansion Series One bundle is truly different. The plug-ins offer good sound quality and a wide range of both conventional and unusual effects. Each user interface is well designed and graphically pleasing. Operation was smooth, and the software never crashed. I found the download and installation to be a breeze, and the price is very reasonable.

The bundle does have a few drawbacks. First, neither the DirectX nor the VST version has any presets, so new users will have a hard time understanding how to set the parameters and grasping the plug-ins' potential. FXpansion told me that the upcoming versions for both DirectX and VST (to be released in June or July) will include presets.

Second, MIDI control does not work in the VST plug-ins. Instead the VST versions rely on VST automation. However, any VST application that can read DirectX plug-ins can use the DirectX versions.

In another unusual quirk, the settings on some plug-ins randomly change on mouse-over. FXpansion is working on the problem. Last, although the plug-ins sound fairly clean, the output unexpectedly overloaded at low settings, creating unintended distortion.

Check out the plug-ins before you buy: screen shots and sound demos are available on the FXpansion Web site. These will give you a pretty good idea of how the plug-ins look and sound, so you can decide if they're right for you.

Overall the bundle offers a great deal of sonic-manipulation capabilities for a very reasonable cost. If you want to alter your sounds in funky and far-out ways, FXpansion Series One is just what you're looking for.

Phil Darg *is an independent composer and producer. His latest work is the MP3.com single "Tough Guys Always Finish First."*

M B H O

MBNM 608 C-L

A bright, versatile mic from a little-known German manufacturer.

By John Ferenzik

Seated snugly in die-cut blue foam rubber inside a black, vinyl-covered storage case, MBHO's MBNM 608 C-L large-diaphragm condenser microphone makes a powerful first impression. The round enclosure that houses the diaphragm is perched, headlike, on a narrow barrel; the mic resembles a lollipop, albeit one with a flat-black Stealth-fighter finish. Cradling the 608 in my hands for the first time, my response was an unbridled "Wow."

The MBNM 608 C-L's list price of $1,299 also makes a strong first impression—that's not exactly chump change. But the 608 does offer three polar patterns—cardioid, omnidirectional, and figure-8—so it's considerably more versatile than any of the single-pattern cardioid condensers that have inundated the market in recent years. On the other hand, the 608 does not provide a highpass filter or an attenuation pad—typical features on multipattern condenser mics. Both the 608's price and its mix of features position it somewhere between the low and high categories, making this mic an attractive option for the upward-leaning personal-studio owner and a great value for the professional engineer.

NEW KID ON THE BLOCK

I used the MBNM 608 C-L for several weeks in a variety of environments and applications. In addition to cutting sessions in my personal studio, the mic clocked time at various other studios, recording vocals, drums, percussion, and an assortment of instrument overdubs. The 608 made its imprint on ADAT, DAT, hard disk (via DAW), and analog tape. I also ran it through a number of mic preamps and processors—including the Avalon VT737 SP, SPL Micman, Focusrite Platinum, dbx 1086, and A.R.T. Pro MPA—as well as the onboard preamps on several mixers.

I took care to be consistent when cutting with the 608: no EQ tweaks were done on initial recording passes, and compression was used sparingly, if at all. When the 608 was pitted against other mics (such as a Neumann U 87 or an AKG C 414), the signal path was duplicated precisely, down to the mic cable.

YOUR RESPONSE, PLEASE

Different polar patterns typically yield different frequency responses, and these are indicated in the literature accompanying the MBNM 608 C-L. Both the omni and cardioid patterns boast impressively wide responses, from 20 kHz down to 10 Hz (cardioid) and 5 Hz (omni). The figure-8 pattern provides a more modest response, ranging from 40 Hz to 18 kHz.

Overall, the 608 tends to sound brighter than the mics that I compared it with, particularly from around 3 kHz up. It also has a slightly hotter output, which necessitated attenuation of the signal at the preamp stage in some instances. As for proximity effect, the 608 retains more low end at greater distances than most of the comparison mics.

PLAINLY SPEAKING

A good measure of any mic is how well it captures spoken-word performances—just ask anyone who records TV or radio spots. With that in mind, I tried the MBNM 608 C-L on several male and female voice-overs, cutting to hard disk via DAW.

On male narration, the 608 was breathtaking. Time after time, it captured mile-wide voices with exceptional presence and detail. And I didn't need to apply EQ after the fact, either. For comparison, I tried identical readings with two popular high-end mics, but both sounded darker and less exciting than the 608. In short, the 608 seized the day.

On female narration, the 608 tended to sound a bit thin, with too much emphasis on the highs above 5 kHz. (In one instance, the same reading through an Audio-Technica AT4047 sounded fuller.) When I switched the polar pattern from cardioid to omni, however, the 608's sound plumped up nicely. In all cases, the omni pattern captured richer vocals and smoothed out the harshness in the high frequencies.

YOU'VE GOT MALE

After the voice-overs, I was eager to see the 608 handle singing chores. First, I tagged along for a jingle-recording session featuring a male vocalist. The 608 performed stunningly. The singer found the mic flattering to his voice and a joy to track with. From the engineer's standpoint, the 608 proved to be a "set and forget" mic, producing perfect tracks without EQ and with only the slightest bit of compression during recording. The clients, too, were happy with the results—in the crucible of the "get it done" jingle session, the 608 showed itself to be *the* mic for male vocals.

I also used the 608 to record some of my own vocals for an upcoming CD. I was not disappointed: when it was my turn to step up to the pop filter and hit a homer, the 608 delivered the goods. I liked how my voice sounded in the headphones while recording, and both the lead tracks and layered background vocals sounded terrific on playback. There was a ton of high-end sparkle, helping my voice cut through the mix—every male singer's dream.

GIRL TROUBLE

The 608 didn't fare as well on female vocalists. At a jingle session, the mic

The MBHO MBNM 608 C-L is a versatile large-diaphragm condenser mic with a bright, articulate sound. It provides three polar patterns but has no bass rolloff or attenuation pad.

For manufacturer information, please see Contact Sheet, p. 205.

was quickly ditched for another large-diaphragm condenser after the vocalist complained that the 608 was too "crispy"-sounding in the headphones. Indeed, the track sounded brittle on playback. Because the clients were present and the clock was ticking, I had no chance to experiment. (Such is the tyranny of the equation "time equals money.")

I tried the 608 on another female vocalist a few days later at a demo session. This time things were more casual, so I was able to experiment. Again, though, the 608 proved to be less than ideal. A pronounced bump between 5 to 8 kHz overshadowed the mids and low frequencies; accordingly, the vocalist felt that her voice sounded too thin. We salvaged the track with EQ, reining in the problematic highs and goosing 200 Hz by 2 dB or so. However, the effort spent tweaking the track left us all feeling a little drained—to paraphrase B. B. King, the thrill was gone. In the end, we recut the track with an AKG C 414, which provided a warmer, fuller sound.

BATTLE OF THE SAXES

At another demo session, I used the 608 to cut both tenor and baritone saxophones (separately) for a song that needed a sax section. While I got usable results by rolling off some high end, the 608 didn't knock me out. As with the female vocalist, the 608 made the tenor sound thin and edgy. The bari fared better—I didn't mind the mic brightening up the lower pitches, but it still made the horn sound harsh in the upper register.

I ended up using a fuller-sounding mic to record the first passes, but, on a hunch, I went back to the 608 for doubling tracks. Here the 608's brightness made the doubles stand out in a way that complemented the original tracks. That quality helped create the illusion of an entire horn section instead of just one guy dubbing away. And because I was miking only one musician, I was able to use the fuller-sounding omni setting on the 608, which helped.

ZING OF STRINGS

The 608 proved an excellent choice for miking acoustic guitars with both steel and nylon strings. Though I had to spend time positioning the mic to locate a spot where it didn't sound too

boomy, the effort paid off handsomely. On steel-string overdubs, the 608 sounded terrific, capturing both the bite (800 Hz to 2 kHz) and airiness (above 10 kHz) critical to acoustic-guitar tracks. There was plenty of warmth down low (200 Hz), as well.

As good as the 608 sounded on steel strings, it was absolutely *spellbinding* on a nylon string guitar. The high end of the darker-timbred classical guitar sparkled, while the overall tone retained its realism and warmth.

As for which polar pattern worked best, it depended on the desired final result. The omni pattern emphasized the low end slightly more than the cardioid pattern, and the figure-8 worked best if I wanted to round off the upper highs and eliminate boominess. Of course, by using those patterns I lost the rejection provided by the cardioid pattern, but with a little effort spent isolating the guitar, I was able to craft killer tracks.

In omni mode, the 608 also proved adequate to the task of recording an upright acoustic bass. On a walking bass line, it captured a focused midrange; full, round low end; and more than enough high end for pizzicato playing. Although the high-end zing—particularly around 3 to 5 kHz—made for a sound more modern than vintage, the bassist on the session remarked that he liked how the 608 captured his instrument's "growl" in the midrange without sounding thin.

Although the 608 did a passable job on a plucked performance, arco was another story. I asked the bassist to bow a few notes, curious about how the 608 would react. Pleasant-sounding bowed-bass tracks are notoriously difficult to capture with condenser mics, especially those that lean toward being bright. Not surprisingly, the 608 sounded a little too scratchy for comfort.

TOTALLY AMPED

It took me a while to find a good spot for the 608 in front of a cranked guitar amp—I've had a much easier time finding that magical position with other mics. With the 608 up close to the grille cloth, small changes in position yielded dramatic shifts in overall timbre, particularly at higher volumes. However, none of these close-to-the-grille positions really worked—the mic was simply overwhelmed by the roar, sounding boomy and bass-heavy.

When I moved the mic 1 or 2 feet back from the cabinet, though, I was rewarded by a variety of tasty tones. Moving the mic got rid of the boominess and brought the guitar's lower mids into focus without losing the high-end bite.

I also got good results when miking the amp from across the room, 15 to 20 feet away. Of course, room ambience played an increasing role the farther back I got, but the 608's "crispness" kept secondary reflections from sounding boxy. I took advantage of the room's acoustics and forged an array of great-sounding guitar tracks.

KEYS OF LIFE

Piano is my main instrument, so I was keen to hear the 608 on my acoustic grand. I set up with a dbx 1086 mic preamp and an ADAT XT20 and proceeded to position the mic, trying it both beneath the grand's propped-up lid and placed in various points around the room. I also auditioned all three polar patterns.

After getting a sound that I liked, I recorded some improvisations in various styles. I also used the mic to overdub piano tracks against reference tracks of a song that I had written and recorded previously.

Although with some strategic positioning the 608 could be used effectively "under the hood," it really excelled when placed a few feet back from the piano. The omni polar pattern, in particular, captured the piano's majesty, beautifully blending the sound with room ambience. The figure-8 pattern also yielded good ambient tracks, with less brightness on top and a tighter-sounding low end.

I got my best results by positioning the 608 outside the piano in omni mode, augmented by a near-coincident pair of small-diaphragm condenser mics suspended above the harp and pointing slightly forward toward the hammers. The deep, realistic ambience of the 608 track combined with the drier stereo image from the two condensers to yield a gorgeous sound.

MBHO
MBNM 608 C-L
large-diaphragm condenser mic
$1,299

AUDIO QUALITY ■■■■■
VALUE ■■■■
1 2 3 4 5

PROS: Exceptional for male vocals and great for voice-overs, acoustic guitar, drum overheads, percussion, and room miking. Compact, lightweight, easily portable. Three polar patterns. Nice storage case.

CONS: Sounds too bright on some applications. No bass rolloff or attenuation pad. No mic clip or shock-mount. Recessed polar-pattern switch is hard to access and verify.

ROAM THE TOME

If there were a book called *All the Ways You Can Mic a Drum Set,* it would probably make the Bible look like a pamphlet. Though I didn't get a chance to audition every conceivable drum-miking technique, I did employ a variety of miking methods on drum sets, usually with stellar results. As the drum tracks made clear, this mic has excellent transient response and a very realistic sound.

As an overhead, the 608 proved to be an ace performer, equally capable of capturing the bright plash of the cymbals, the detail of the ride-stick work, and the punch and timbre of the toms. When compared with a popular large-diaphragm mic in this application, the 608, impressively, sounded both brighter and fatter. Whether as an overhead or a room mic, the 608 would be among my first choices for capturing the raw energy of a drummer thrashing the kit.

As a close drum mic, the 608 again yielded aural gold. I especially liked it on hi-hat, where it did a great job of capturing the attack of the sticks on the closed cymbals as well as the nice midrange "chick" sound made by closing the cymbals with the foot. The 608 sounded beautiful, too, on some snare drum brush work.

Capturing backbeats and rim shots from the snare was trickier due to the 608's hot output. But with careful mic positioning and a little peak limiting, I tamed the spikes and got nice, fat snare tracks.

The 608's cardioid polar pattern is not the tightest available, so finding the position that best minimized leakage from the rest of the kit took some work. Even then, I had to spend more time setting a gate to sufficiently de-emphasize the bleed from adjacent sources—not atypical for a condenser mic on snare, of course, but worth noting.

PERCUSSION, PERCHANCE?

The 608 scored high marks on percussion, too. Positioned close and set to cardioid, the mic captured plenty of attack and high end from both congas and bongos without sacrificing the drums' resonance. When moved back a bit and flipped to omni to catch some room sound, the 608 really impressed me—both sets of drums retained their snap and resonance without being muddied by the darker sound of the room.

The 608 did justice to most other small-percussion overdubs—claves, shakers, cabasa, and maracas—especially in capturing the high end and in helping the tracks cut through the mix. A notable exception was on

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tambourine, where the 608 proved too harsh—it overexposed the instrument's "jangly" character, sounding cold and ultimately rendering the track unusable.

OTHERS AND DRUTHERS

I also took advantage of the 608 to record a few out-there samples for my personal sound-design library. On large tubular wind chimes, the 608 accurately caught the clangorous attack and hollow ringing of the copper tubes. The trickling of a large rain stick also benefited from the bright cast that the 608 imparted to the sample.

Using the 608 with an A.R.T. Tube MP mic preamp cranked to 10, I created an intentionally distorted voice-over for a special effect. Here, the 608's hot output and cutting high end served to overdrive the preamp's tube for a warm and fuzzy approximation of an old TV broadcast—perfect for the application that I had in mind. Sure, I could have created this effect any number of ways—for instance, by patching in an effects unit and notching the EQ on a pristine vocal track—but the 608's convenient size and bright tonal characteristics offered a quick, one-step solution.

FINAL VOTE

While there will always be the bittersweet ache that comes from wanting *the* legendary large-diaphragm condenser mic (which costs as much as the car you own), it's nice to know that a sensibly priced option like the MBHO MBNM 608 C-L occasionally comes along. For male vocals especially, the 608 is a stunner—it stood its ground against some steeply priced (and steeped-in-legend) competition. The mic was also a first-rate performer on acoustic guitar, drums, and percussion, and as a room mic.

I like having the cardioid, omni, and figure-8 polar patterns available; they make the 608 quite a flexible player. Both the omni and figure-8 settings are great for capturing room ambience. I also like the different frequency responses made available by the multiple polar patterns—having a palette of tonal options to play with is a plus.

The 608 gets high marks for convenience. For one thing, its narrow barrel eliminates the need for a special (read costly) clip or shock-mount. Indeed, I

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MBNM 608 C-L Specifications

Type	side-address, external-DC polarized ("true" condenser), pressure-gradient transducer
Capsule	1", gold-sputtered, dual-diaphragm
Polar Patterns	cardioid, omnidirectional, figure-8 (switchable)
Frequency Response (± 1.5 dB)	cardioid: 10 Hz–20 kHz; figure-8: 40 Hz–18 kHz; omni: 5 Hz–20 kHz;
Dynamic Range	120 dB
Signal-to-Noise Ratio	cardioid: 80 dBA; figure-8: 81 dBA; omni: 79 dBA
Self-Noise	cardioid: 21 dB; figure-8: 20 dB; omni: 22 dB
Maximum SPL (for 0.5% THD @1 kΩ)	cardioid: 132 dB; figure-8: 132 dB; omni: 133 dB
Sensitivity (@1 kHz)	cardioid: 11 mV/Pa; figure-8: 8 mV/Pa; omni: 10 mV/Pa
Highpass Filter	none
Attenuation Pad	none
Power Requirement	48V phantom
Dimensions	7.88" (L) × 2" (W) × 1" (D)
Weight	8.5 oz.

was able to get by with the mic clip from a small-diaphragm condenser mic in my collection—a good thing, because the 608 ships without a mic clip or shock-mount. I also appreciate the slim profile and light weight of the 608. This mic is easy to position in tight spaces, making it a sensible choice for location recording, too.

I do have a quibble with the 608's polar-pattern selector. The selector is small and deeply recessed—I had to use the tip of a ballpoint pen to change patterns—and I had some difficulty verifying which pattern it was set to. In the mic's defense, the hard-to-reset switch makes the 608 less prone to unintentional pattern changes during handling and positioning. Moreover, although I was initially apprehensive about the 608's lack of a low-end rolloff or attenuation pad, I barely noticed their absence while using the mic.

Given its tendency to sound too bright on some applications, the MBHO MBNM 608 C-L is not the ideal candidate for every occasion—but then, what mic is? I still found it to be incredibly versatile, and I give it a ringing endorsement.

John Ferenzik *is a multi-instrumentalist who has played with Todd Rundgren and other noted artists. He recently completed a solo project*—Zero Points for Zeus—*which was released in May. You can e-mail John at frnzk@pond.com or visit his Web site at www.ferenzik.com.*

recording right away. The Mixer allows you to switch between line-level and mic-level input gains.

WIELDING THE UNWIELDY

At first I was taken aback by the VXpocket's bulky XLR connectors. Size and weight are critical factors in notebooks and their accessories, so why burden your portable setup with a clunky cable cluster? On the input side, the rationale is obvious. XLRs let you plug high-quality mics right into the card.

On the output side, though, the logic is less clear. I like being able to plug my notebook right into my mixer's balanced inputs, but in the field the absence of a headphone output is frustrating. I don't want to drag along a headphone amp for mixing or editing by the pool or during a long flight.

APP PERPLEXING

The Mixer is an unassuming but powerful (and occasionally puzzling) applet essential for controlling the card properly (see **Fig. 1**). It doesn't install automatically, but you can set it to load at startup if you wish. The manual says nothing about the Mixer, but the Mixer's short Help file tells you most of what you need to know. The Help file's only weak area is its description of how to set input levels.

Input levels are controlled with the Mixer's Analog In slider and two numerical parameters, Nominal In and Headroom In, which are set in the Advanced Input dialog box. Together the two settings are the equivalent of a mixer's input-trim pot. For example, to record a live performance from a mixer's stereo line out, you'd set Nominal In to the mixer's line-level output (typically –10 dBV or +4 dBu). This lets the VXpocket know what incoming level to expect. Next you'd set a Headroom In value reflecting how hard you expect to push that line out. The VXpocket deducts this value from the input gain to give you a margin of error before clipping occurs. After setting Nominal In and Headroom In, you'd make fine adjustments with the Analog In slider.

VXpocket
Minimum System Requirements
Mac: PowerBook 2400; 32 MB RAM; Mac OS 8.1; QuickTime 2.5
PC: Pentium/200; 32 MB RAM; Windows 95/98/2000/NT 4.0

To set the VXpocket for mic-level inputs, Digigram recommends starting with a Nominal In value of –27.0 and a Headroom In value of 0.0. If you need more gain, you must choose a lower Nominal In value; –30, for instance. If this seems counterintuitive, just remember that you're telling the VXpocket that the mic's output is lower and therefore needs more boost.

If you're importing digital tracks via the S/PDIF connections, set Nominal In and Headroom In to 0.0 and center the Analog In and Digital In faders to get an accurate digital clone of the source. When you export digital material, you select Consumer (S/PDIF) or

WHAT DO THEY DO FOR AN ENCORE?

Perhaps out of wishful thinking, a number of people have hailed the VXpocket's cousins in the PCXpocket series as viable options for mainstream music production. I made the same error on my Web site prior to the introduction of the VXpocket. Similar hopes were pinned on two recently released multichannel versions of the PCXpocket.

Unfortunately, the PCXpockets don't support popular, commercially available music software. The PCXpockets are special-purpose devices intended for testing, measuring, and industrial applications, and they include onboard processing for which custom applications must be written, often by third-party software developers. Moreover, a PCXpocket costs twice as much as the VXpocket.

The good news, however, is that Digigram has just announced two new VXpocket models, of which one, the VXpocket 440, is the first PC Card to support four input channels at once. The 440 brings multichannel audio to notebook recording, offering four balanced inputs and four balanced outputs on XLR connectors. It features the same 24-bit converters as its stereo sibling, as well as S/PDIF digital I/O. In another first for laptops, Digigram has added SMPTE (LTC) time-code input, a move that will certainly be welcomed by anyone doing video work in the field. The 440 should be available in July for $1,069.

Due to the bandwidth limitations inherent in PC Cards, the 440 doesn't offer full-duplex recording on four inputs and four outputs. When it's operating in Full-Duplex mode, you get only one stereo input and one stereo output, as with the original VXpocket. To get four simultaneous inputs or two simultaneous stereo inputs, you need to shut down the outputs completely. You won't be able to monitor previously recorded tracks, but you can monitor all four inputs in real time just as you can with the VXpocket. Similarly, to access all four outputs at once you must give up on the idea of recording at the same time.

At the time of this writing, Digigram was planning the VXpocket v2 upgrade, which adds SMPTE (LTC) time-code input to the original complement of stereo analog and digital I/O. Digigram built time code into the card's original design, so owners need only purchase a new cable for $75 to employ the upgrade feature. Version 2 will sell for the same price as the VXpocket ($729).

It's important to note that applications requiring Sound Manager or the Microsoft Wave protocol can't use the LTC input because those two standards don't support time code. To take advantage of the new VXpockets' synchronization capabilities, you must use either an ASIO-compatible application on the Mac or PC or a DirectSound-compatible application on the PC.

Having seen so many PC Cards abandoned or discontinued, I'm impressed by Digigram's accomplishments in the VXpocket series. Not only has the company succeeded where others have failed, it has done so with attention to detail and an ear for high-quality audio.

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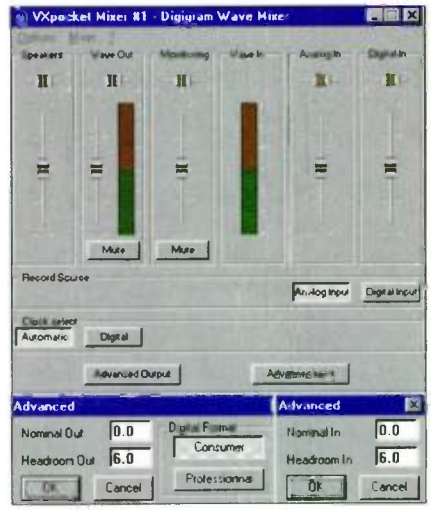

FIG. 1: The VXpocket Mixer gives you access to all of the VXpocket's important features. In addition to giving you control over levels for recording, monitoring, and playback, it enables you to select line- or mic-level gain for the inputs through the Advanced Input dialog box.

Professional (AES/EBU) format in the Advanced Output dialog box.

The Mixer gives you control over one of the card's most valuable features: a direct monitor loop. This loop feeds the incoming signal directly to the output for real-time monitoring. The incoming signal bypasses your software, so it won't suffer from latency, but the bad news is that you can't apply plug-in effects. A fader and mute button let you adjust the monitor signal without affecting recording levels.

COAXIAL COAXING

After spending years coaxing better audio out of my notebook than it was ready to give, I had a great time using the VXpocket. It sounds good and is easy to use. Its mic- or line-level operation makes it extremely flexible for field recording. I tried plugging a Shure SM58 mic directly into it and running a condenser mic through a tiny tube preamp, and the VXpocket happily and accurately recorded the results in both instances. Its real-time monitoring function made overdubbing tracks easy and pleasant.

The only glitch I encountered during my tests was an occasional dropout error in Cakewalk's *Pro Audio* 9, and that happened so seldom that I don't know whether to pin the blame on the VXpocket or another variable. If you're looking to turn your notebook into a serious DAW, give the VXpocket a listen.

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BEST SERVICE

Xsample

An unorthodox collection of sampled instruments.

By Rob Shirak

Best Service took an uncommon approach when creating the nine-CD Xsample sound library: instead of cramming hundreds of samples into the available space, it concentrated on a few dozen instruments sampled extensively. The results are on the whole sonically outstanding, although some glaring gaps in the content may leave you scratching your head.

The Xsample library (available in Akai-compatible, E-mu EOS, and GigaSampler formats) is a set of nine audio discs; each one contains well over 500 MB of samples (see the table "The Xsample Sound Library"). Some discs contain only three or four instruments; obviously, a lot of sample memory was devoted to capturing the numerous long multisamples and variety of articulations. Many instruments have individual programs for several velocities (forte, mezzopiano, and so forth)—sometimes as many as four levels—in addition to larger programs that Velocity-switch between the layers.

Practically all of Xsample's instruments are available in both stereo and mono, and some programs include appropriate chorused versions. A few instruments (the Rhodes, for instance) have variations resampled at a lower rate to conserve memory. The Akai and E-mu versions of Xsample contain additional programs that use only four samples per octave (at full sample frequency), reducing the memory requirement by roughly one third. I reviewed Xsample on an Akai S6000 with 128 MB of memory.

The Xsample Sound Library

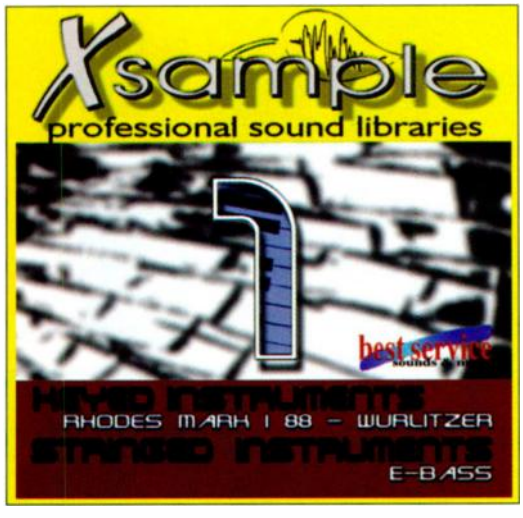
Disc	Instrument
1	Rhodes Mark I 88, Wurlitzer, electric bass
2	alto saxophone, bass flute, 8-string guitar
3	French horn, trumpet, trombone
4	vibraphone, marimba, xylophone, glockenspiel, crotales, steel drum
5	violin, viola, cello, double bass
6	oboe, English horn, bassoon, contrabassoon
7	oboe d'amore, cello (sustained), spinet, clavichord, kalimba
8	celesta, klavicymbel, concert harp, psalter
9	assorted percussion (bongos, congas, drums, water gong, chimes, and so on), special instruments (kazoo, jaw harp), and vocal effects

SOUNDS LIKE . . .

Some of the instruments in this collection are absolutely stellar. Most of the samples were recorded without much room ambience, but despite being dry, they don't sound claustrophobic or constrained.

The Rhodes Mark I 88 on disc 1 is among the finest electric piano samples available. The four-layer, Velocity-switched program is 64 MB on the Akai; the E-mu and NemeSys GigaSampler versions are 81 MB and 218 MB, respectively. If you like the sound of this particular Rhodes, you'll love the realistic multisample, regardless of your sampler format.

The bass flute on disc 2 also sounds incredible and is a rare find in sample libraries. The glockenspiel is exceptional as well, with just the right balance of top end and low frequencies, sounding bright yet natural. Other standouts include the oboe, contrabassoon, celesta, harp, Wurlitzer electric piano, most of the percussion, and the entire collection of solo brass.



The Xsample sound library offers high-quality samples of an eclectic variety of instruments.

COLOR MY WORLD

There's a good-sounding electric bass on disc 1. However, because the programs are all derived from a single instrument, this particular bass sound has a lack of tonal variation.

The instrument selection isn't complete enough to serve as a basic library of traditional sounds. Rather, the charm of this collection is in the sounds of the specific instruments themselves. For example, the collection routinely omits common articulations and groupings—no regular flutes, clarinets, timpani, or ensemble samples of strings, woodwinds, and brass are to be heard. Some of the instruments on Xsample are welcome but unusual, like the oboe d'amore, psalter, 8-string guitar, clavichord, and jaw harp. Considering that there are well over 4.5 GB of samples in the Xsample library, I would like to see a few more bread-and-butter instruments.

Despite the large amount of memory allotted to them, some instruments suffer in their most common incarnations. The forte vibraphone, marimba, and xylophone samples sound splatty and really stand out from the mezzoforte samples in Velocity-switched programs. This makes it difficult to create musically fluid passages. On the other hand, the solo woodwinds concentrate a bit too much on specialty sounds like clacks, slaps, and multiphonics.

Apart from some special-effects sounds, only highly stylized articulations like pizzicato, spiccato, and col legno are available for the solo strings. Xsample lacks the standard legato tones necessary for most melodic and

For manufacturer information, please see Contact Sheet, p. 205.

BEST SERVICE
Xsample sample CD library
nine-CD set $895
individual CD $149.95

QUALITY OF SAMPLES ■■■■
QUALITY OF PROGRAMMING ■■■■
DOCUMENTATION ■■
VALUE ■■■■
1 2 3 4 5

PROS: Contains uncommon instruments and articulations, long samples, and lots of multisamples per instrument. Some samples come in reduced versions. Most samples are available in both stereo and mono versions.

CONS: Many common instruments and traditional articulations have been overlooked.

harmonic work. Of course, you could mix these sounds with common ones from other libraries, but you might lose the continuity between timbres.

Maybe it was Best Service's intention to complement the resources found in more traditional libraries. If that's the case, I would like to see it release another nine-disc set that fills in the holes of the sometimes brilliant Xsample library. The recording quality is uniformly superb; the sound designers obviously know how to sample instruments quite well. I only question some of their instrument and articulation choices.

WHAT'S ON THE PROGRAM

For the Akai version of Xsample, the S2000 format was chosen so that the samples would work with every Akai sampler. To maximize compatibility with all models, Best Service kept Xsample's programming to a minimum. I did notice some inconsistency in the assignment of the modulation wheel and pitch-bend controllers throughout the library, but nothing too problematic.

I would like more programming added to the samples. Much can be done with filtering and envelopes to expand the sonic palette into more adventurous territory. Instead, Xsample simply presents an instrument and its articulation in a raw and naked form.

The documentation provides good information, but the layout makes it difficult to decipher until you've memorized the naming conventions. After learning the coding scheme, you can better determine which programs are stereo or mono, chorused, resampled, and so on, as well as the number of Velocity layers provided.

Xsample proves that sampled instruments are capable of great expressiveness when you devote large amounts of memory to them. This library is all about exploring the depths of specific instruments, and here it largely succeeds. If you like the instruments provided, you'll love the quality of the samples found on these discs.

Producer and composer **Rob Shirak** *is the music director for Burt Bacharach. He has also worked with Dionne Warwick, Elvis Costello, LeAnn Rimes, Stevie Wonder, Mikaila, Luther Vandross, and a host of other artists.*

QUICK PICKS

FUTURITY

Metropolis Science Fiction Toolkit

By Jeff Obee

Futurity's *Metropolis Science Fiction Toolkit* ($99) is a new collection of 16-bit, 44.1 kHz sound effects on audio CD. Upon listening to the first selection, Metallic Whoosh/Long, I wondered if I was hearing *Star Trek: Voyager,* because this track would certainly fit well in that show.

Rushing water, laser zaps, and antique automobile engines are among the sound tools found in Futurity's *Metropolis Science Fiction Toolkit* sample CD.

Future Sounds

All of the CD's sounds are named for their sonic characteristics. For instance, Metallic Shimmer is a high-pitched, heavily processed effect that has a glistening quality. Similarly, two versions of Storm Release (again, heavily processed) sound like water racing through a large tube.

Futurity chose to cover considerable aural territory in a distinct and varied palette: deep harmonic booms, ethereal bell accents, flange/distortion sweeps, aggressive synth warbles, sirens, random holds, lasers, buzzes, screeches, bursts, hits, ambient pads, and much more. Many of the CD's 99 tracks include a few variations on a given effect, so in fact you get 188 sounds. The tracks range in duration from 1 second to 1 minute and 14 seconds; most tracks are short cues that are 1 to 5 seconds long. Many tracks do not have a definite pitch, and therefore they work best as effects. Some tracks do have a pitch and could be used as musical cues or washes.

I like the two Clean Ambient Wash tracks (one lasts 9 seconds, the other 19). Both have a rich, glossy ambience and a glassy vocal texture, not unlike the effect you hear when a person is beamed up in '90s *Star Trek* episodes. Both versions of Low Pulsing Ambience are striking for their resonant, eerie qualities; one offers a cool sample-and-hold element.

The three Noise Impact sounds on track 75 are very effective, as are the three Alert Siren cues. Metal Sweep Ambience sounds like a gargantuan, pulsing cymbal or gong and creates a broad, sustained backdrop. Low Voice Detuning is especially good—a long, deep, spooky effect with synth chords that drift in pitch.

Sci-Fi Tools

The sound sources for the disc's effects include guitars, water, metal, and machines—such as a 1912 Pierce Arrow and a 1911 Stanley Steamer automobile—as well as a stationary multipurpose engine that was often used in agriculture in the early 20th century. These sound sources were recorded with a Crown SASS-P stereo PCM microphone into an HHB Portadat. On the lo-fi side of things, some old 12-bit Sequential Circuits Prophet-2000 samples were recorded with a Shure SM57 microphone and imported into Digidesign's Pro Tools.

The source sounds were processed with Eventide H3000 and Lexicon 480L and MPX 1 processors along with some Roland synth modules, a Prophet-5, and a variety of '70s and '80s stompboxes. Symbolic Sound's Kyma, Digidesign's Pro Tools/24 Mixplus, BIAS's *SFX Machine,* and Waves' TDM Bundle were also used. The effects are applied tastefully, but I would like to have additional versions with no reverb or extraneous processing so that I could apply my own.

Future Thoughts

The documentation for *Metropolis Science Fiction Toolkit* is sparse. The effects are listed on the back of the jewel case by track number, description, and duration, and are continued inside the single-page front sleeve. Aside from company information, copyright warnings, and a few thank-you credits, that's all you get—but that's really all you need to make use of this disc.

The collection is on the expensive side—$99 is a little steep for an audio CD. Using *Metropolis Science Fiction Toolkit* with a sampler or importing its tracks into your DAW is more of a hassle than it would be if this were a CD-ROM. That notwithstanding, the sound effects here are all well crafted, and some are superb. All of them would work well as effects for sci-fi, horror, or action pieces.

Metropolis Science Fiction Toolkit's producers obtained the Kyma sound-design system toward the end of the CD's production, but they were unable to tell me specifically which sounds they used. I've never heard Kyma, but this program won an **EM** Editors' Choice Award (see "1998 Editors' Choice Awards" in the January 1998 issue of **EM**), and I'm told that it will be used extensively on Futurity's next disc.

You can hear sample selections from *Metropolis Science Fiction Toolkit,* as well as other products, on Futurity's Web site. If you need post-production sound effects for film or video, or if you like having esoteric sounds in your library, I recommend perusing these samples. You may just find the effects cues that you're looking for.

Overall EM Rating (1 through 5): 3.5

MICROBOARDS

StartRec 400

By Alex Artaud

It wasn't so long ago that burning your own CD meant shelling out a huge bundle of dough. Nowadays, you can opt for either a stand-alone CD recorder or one that acts as a computer peripheral and get away with spending between $300 and $700. And with CD-R media costing as little as a buck apiece, it's hard to imagine a better way for the budget-conscious recording artist to go. But what if you want to burn more than one disc at a time? And what if you want to burn them quickly?

Enter MicroBoards Technology's StartRec 400 ($3,995), a CD-duplication system that includes a 6 GB hard disk, helpful editing features, and multiple burners that write at up to 8× speed. And while CD-to-CD duplication is possible, you can also transfer

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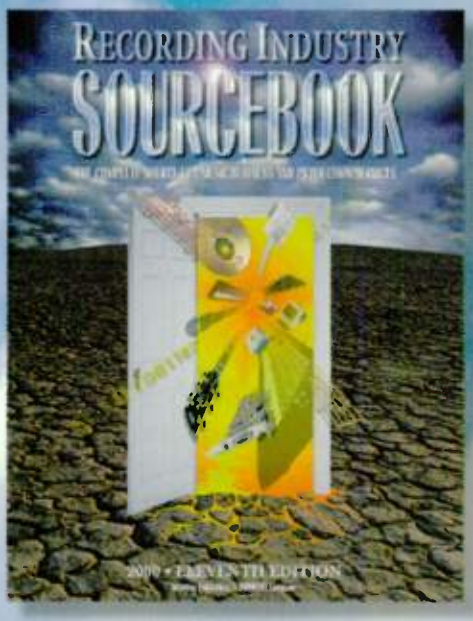

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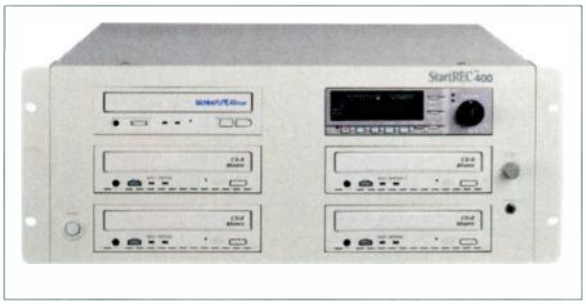
audio (or data) onto a hard disk, organize it, and then burn CDs. The price for such convenience may seem steep, but this is a fast system that could easily win over artists who run homespun labels, as well as studios in need of an efficient backup system.

Lay of the Land

The StartRec 400's layout is logically organized. The unit comes with one CD source deck and four CD recorders that feature stable Plextor drive mechanisms. The control panel, located in the upper right corner of the box, sports a small LCD and controls for navigation and record level. There are buttons for transport functions such as record, playback, pause, fast-forward, and rewind; some buttons perform double duty—for instance, selecting menus as well as entering parameter values. An Input button lets you choose either analog or digital inputs, and status LEDs for each input are provided. The front panel also features a ¼-inch headphone jack and a level knob.



The MicroBoards StartRec 400 is a convenient, powerful option for anyone who needs a fast CD-duplication system.

The rear panel offers analog inputs and outputs via balanced XLR and unbalanced RCA plugs. Digital connections include AES/EBU and coaxial and optical S/PDIF. A 9-pin port for a remote control is also provided, but it isn't mentioned in the manual. These features all fit into a 4U rackspace and are housed in a sturdy chassis.

Start Your Burners

The StartRec 400 is organized into three sections (or menus) for copying, extracting tracks from CDs, and recording and editing audio. Each function is straightforward, but it takes a while to get used to navigating the unit. Part of the problem is a poorly organized manual that isn't much better than a flowchart. A lucid description of the unit's operations, complete with step-by-step diagrams, would be much more helpful.

Once I became familiar with the system's nomenclature, I was able to operate it painlessly. I copied from CD to CD, CD to hard disk, and hard disk to CD. I also took several CDs, marked and extracted some tracks for a custom job, and encountered no glitches on the finished disc. Additionally, the StartRec 400 records seamlessly to hard disk from outside sources, a feature not seen in most duplication systems.

I found some other cool features, as well. An audio-verification function lets you compare each duplicated file with the original to help ensure accurate digital audio transfers. Within the Audio menu, you can automate the placement of a track marker (a *start ID* in DAT lingo) by setting the desired decibel level below which an index will be recorded. Working with audio in the hard disk affords you the most flexibility, letting you divide one track into two, remove an index to merge two tracks together, or move a track to a different location before burning.

You can quickly erase data from any of the three partitioned areas on the hard disk (which measure 2 GB each). Be careful, though, as there is no onscreen warning that you're about to torch your work: once you press OK, it's gone. Also, the StartRec 400 ignores SCMS data on CDs and leaves it up to you to determine how many times the CDs that you burn can be duplicated.

A Fast Burn

If you work briskly using four burners, you should be able to knock out between 25 and 30 CDs, each containing 60 minutes of material, within an hour. Other relatively inexpensive CD duplicating systems are available, but few if any of them combine all of the features found in the StartRec 400. Anyone who is serious about CD duplication should check out this system.

Overall EM Rating (1 through 5): 4

F7 SOUND AND VISION

Concept: FX2 (Mac/Win)

By Jeff Obee

If you have high-quality, portable recording gear and lots of processors, you can pluck sounds out of the environment and design your own audio creations. However, those of us without the necessary equipment must rely on sound-effects CDs to fill our sonic palette. Fortunately, F7 Sound and Vision has decided to help us expand our options into some uncharted aural territory with its new *Concept: FX2* CD-ROM (Mac/Win; $69.95; bundled with *Concept: FX*, volume 1, $99.95). It offers 305 unusual stereo sound effects in 16-bit, 44.1 kHz AIFF and WAV formats.

Concept Creation

Concept: FX2 bears some resemblance to Rarefaction's *Digital Dysfunctions*, in that a variety of intriguing sound sources were used as raw material. (For a review of *Digital Dysfunctions*, see the April 2000 issue of **EM**.) The sounds were then processed beyond recognition (often run through multiple processes) and given amusing names.

Producer Michael Oster employed a surprising array of resources to create these sounds, including a live cicada, an early '50s Telefunken radio, a huge carbon-steel exhaust pipe, a metal door, kitchen utensils, ARP Odyssey and Roland JP-8000 synths, nature sounds, machines, city ambiences, object impacts, and a dying Mac IIci. In other words, anything that sparked his imagination was fair game.

The basic sounds were then "sliced, diced, and rearranged" in Pro Tools and processed extensively with AudioSuite plug-ins (pitch-shifting, time compression/expansion, EQ, dynamics compression, gating, and gain) and TDM plug-ins (Digidesign's *D-Fi* series, Focusrite's *D2* and *D3*, *DINR*, TC Works' *MegaReverb*, and TC Works' *MasterX*). Limited use was also made of Digidesign's *Sound Designer II* and Tom Erbe's *SoundHack*.

Occasionally the signal was routed in series through several channels of Mackie EQ (boosting the frequency with a tight Q), then fed through a Roland SDE-1000 delay with the feedback set to 100 percent.

Sounds Like a Concept to Me

And what was the result of all this bizarre audio manipulation? A veritable treasure trove of fascinating, one-of-a-kind sound effects. Some are short bursts of sound in the 20 Hz to 200 kHz range; others are alien textures and ambiences that last several seconds. The lengthiest recording is almost 2 minutes and consumes more than 19 MB of disk space. Most of the sounds, however, range from less than a second to about half a minute. None of the sounds has a definite pitch per se, so they're less useful as instrumental samples than

F7 Sound and Vision's *Concept: FX2* CD-ROM offers hundreds of unusual and heavily processed sound effects in WAV and AIFF format.

straight-ahead sound effects. Some of the longer pieces have a distinctly rhythmic element, making them suitable as loops for rhythm tracks. And several of the shorter offerings have the potential to be unique percussion sounds.

Among the short effects, I especially like TrampoSprings, with its heavily processed metal-spring texture. Another favorite is UnnaturalSplash, which sounds like a single wave hitting a hard surface. LarvalGasDischarge2 is an airy, gurgling emission sound that definitely justifies its name. Many of the shorter sounds have a lot of presence, which helps them stand out in a mix. WetGutPunch, for example, hits with a solid crunch and could be used as an interesting snare drum substitute.

I particularly enjoy LophResonil, a 1 MB file with a rising and falling resonance combined with a rhythmic squishy sound. When cut and looped, it makes an unusual percussive bed. ZappySonnetRhyme is another medium-length chunk; its churning, rhythmic quality also lends itself well to use as a percussion loop.

Of the longer sounds, BL239 stands out with its edgy, distinctive character. At 4.4 MB, it rolls along for well over 20 seconds, sounding somewhat like an object being shaken inside a small plastic container. I snipped and looped bits of the sound for rhythmic effects. HumpbackInterpretation conjures up visions of whales in space; and if hoary monsters are your bag, you'll love IntoTheMurk—just add a dark swamp and fog. The 5 MB LotsaBirds lives up to its name by offering 30 seconds of cacophonous, high-pitched, birdlike chatter.

Concept Critique

Concept: FX2 gets high marks for value. At just under $70, this CD is an outstanding buy for those times when you want to add the occasional strange bit of sound to your sonic soup.

Because this assortment is in CD-ROM format, importing the sounds directly into your favorite audio-editing program is a piece of cake. And the files are provided in both AIFF and WAV format, so you can use them not only in Macs and PCs, but also in the new Kurzweil and Akai samplers. The audio quality here is excellent and the price is right—this disc has definitely found a home in my sound-effects CD collection. Check out the F7 Sound and Vision Web site, listen to the demos, and see if *Concept: FX2* will find a home in yours.

Overall EM Rating (1 through 5): 4

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EM Advertiser Sales Regions and Representatives

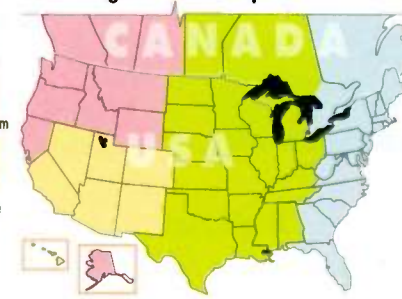
Joanne Zola, Northwest
(510) 653-3307
joanne_zola@intertec.com

Mari Stancati, Southwest
(323) 845-1881
mari_stancati@intertec.com

Julie Clark, Midwest
(616) 687-8848
julie_clark@intertec.com

Joe Perry, East Coast/Europe
(770) 343-9978
joe_perry@intertec.com

Jef Linson, Classifieds
(800) 544-5530
jef_linson@intertec.com



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Event Electronics tel. (805) 566-7777; fax (805) 566-7781; e-mail info@event1.com; Web www.event1.com

Genelec tel. (508) 652-0900; fax (508) 652-0909; e-mail genelec@compuserve.com; Web www.genelec.com

JBL International tel. (800) 852-5776 or (818) 895-3496; fax (818) 884-2974; Web www.jblpro.com

Vergence Technology, Inc. tel. (877) 464-8776 or (707) 751-0270; fax (707) 751-0271; e-mail customerservice@vergenceaudio.com; Web www.vergenceaudio.com

Yamaha Corporation of America tel. (714) 522-9011; fax (714) 739-2680; e-mail info@yamaha.com; Web www.yamaha.com

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Alesis Corporation tel. (800) 525-3747 or (310) 255-3400; fax (310) 255-3401; e-mail info@alesis.com; Web www.alesis.com

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BBE Sound tel. (800) 233-8346 or (714) 897-6766; fax (714) 896-0736; Web www.bbesound.com

Behringer USA Inc. tel. (425) 672-0816; fax (425) 673-7647; e-mail support@behringer.com; Web www.behringer.com

Coleman Audio tel. (516) 334-7109; fax (516) 334-1420; e-mail coley@colemanaudio.com; Web www.colemanaudio.com

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E-mu/Ensoniq tel. (831) 438-1921; fax (831) 438-8612; e-mail info@emu.com; Web www.emu-ensoniq.com

Fatar/Music Industries Corp. (distributor) tel. (800) 431-6699 or (516) 352-4110; fax (516) 352-0754; e-mail mic@musicindustries.com; Web www.musicindustries.com

Glyph Technologies tel. (800) 335-0345 or (607) 275-0345; fax (607) 275-9464; e-mail info@glyphtech.com; Web www.glyphtech.com

MOTU (Mark of the Unicorn, Inc.) tel. (617) 576-2760; fax (617) 576-3609; e-mail info@motu.com; Web www.motu.com

Novation USA tel. (888) 782-3166 or 44-162-882-8880; fax 44-162-882-8881; e-mail sales.novation@nova-usa.com; Web www.nova-uk.com

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Best Service/East West (distributor) tel. (800) 833-8339 or (212) 541-7221; fax (212) 541-7015; e-mail sales@eastwestsounds.com; Web www.bestservice.de

Digidesign tel. (800) 333-2137 or (650) 842-7900; fax (650) 842-7999; e-mail prodinfo@digidesign.com; Web www.digidesign.com

Digigram, Inc. tel. (703) 875-9100; fax (703) 875-9161; e-mail input@digigram.com; Web www.digigram.com

Ego Systems, Inc./Thinkware (distributor) tel. (800) 369-6191 or (415) 777-9876; fax (415) 777-2972; e-mail webmaster@egosys.net; Web www.egosys.net

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

Futurity tel. (408) 354-4432; Web www.futurityfx.com

FXpansion tel. 44-7808-157-967; e-mail info@fxpansion.com; Web www.fxpansion.com

Korg USA, Inc. tel. (516) 333-9100; fax (516) 333-9108; e-mail product_support@korgusa.com; Web www.korg.com

Kurzweil Music Systems, Inc./Young Chang tel. (800) 421-9846 or (253) 589-3200; fax (253) 983-8206; Web www.youngchang.com/kurzweil

MBHO/MTC (Music Trade Center) (distributor) tel. (718) 963-2777; fax (718) 302-4890; e-mail info@mbho.de; Web www.mbho.de

MicroBoards Technology tel. (800) 646-8881 or (952) 556-1600; fax (952) 566-1620; e-mail support@microboards.com; Web www.microboards.com

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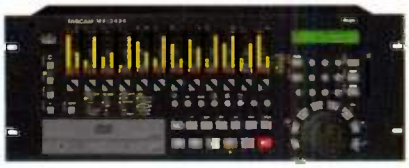
DIGITAL MULTI-TRACK RECORDERS

TASCAM MX-2424 24-Bit 24-Track Hard Disk Recorder



Co-designed by TASCAM and TimeLine Inc., the MX-2424 is an affordable 24-bit, 24-track hard disk recorder that also has the editing power of a digital audio workstation. A 9GB internal hard drive comes standard as well as a SCSI Wide port that supports external LVD (Low Voltage Drives) hard drives from up to 40 feet away. An optional analog and several digital I/O cards are available so the MX-2424 can be configured to suit your work environment. SMPTE synchronization, Word Clock, MIDI Time Code and MIDI Machine Control are all built in for seamless integration into any studio

- Records 24 tracks of 24-bit audio at 44.1 or 48 kHz, or 12 tracks at 88.2 or 96 kHz. Up to 24 tracks can be recorded simultaneously using any combination of digital and analog I/O.
- Supplied 9GB internal drive allows 45 minutes of audio across all 24 tracks
- Wide SCSI port on the back panel allows you to add multiple drives. A front 5-1/2" bay available for installing an additional drive, or an approved DVD-RAM drive for back-up.
- ViewNet MX, a Java-based software suite for Mac and PC offers DAW style editing of audio regions, dedicated system set-up screens that make set-up quicker and easier and track load screens that make virtual track management a snap. Connects to a computer via a standard Ethernet line.
- Can record to Mac (SDII) or PC (.WAV) formatted drives, allowing later export to the computer. The Open TL format allows compatible software to recognize virtual tracks without have to load, reposition and trim each digital file.

Transport Controls-

- Jog/scrub wheel
- MIDI In, Out, and Thru ports are built-in for MIDI Machine Control.

Editing-

- Built-in editing capabilities include cut, copy, paste, split and ripple or overwrite
- 100 levels of undo
- Supports destructive loop recording and non-destructive loop recording which continuously records new takes without erasing the previous version.

Build-In Synchronization-

- TBUS protocol can sample-accurately lock 32 machines together for 384 tracks at 96kHz, or 768 tracks at 48kHz
- Can generate or chase SMPTE timecode or MIDI Time Code.
- Word Clock In, Out, and Thru ports

I/O Options-

- Optional analog and digital cards all provide 24 channels of I/O. There is one slot for analog and one for digital.
- IF-TD24- T/DIF module
- IF-AD24- ADAT Lightpipe module
- IF-AE24- AES/EBU module
- IF- AN24- A-D, D-A I/O module with DB-25 connectors

Software Updates-

- System updates are made available through a front panel Smart Card slot or via computer directly from the TASCAM web site.

DA-78HR Modular Digital Multitrack

The DA-78HR is the first true 24-bit tape-based 8-track modular digital multitrack recorder. Based on the DTRS (Digital Tape Recording System) it provides up to 108 minutes of pristine 24-bit or 16-bit digital audio on a single 120 Hi-8 video tape. Designed for project and commercial recording studios as well as video post and field production, the DA-78HR offers a host of standard features including built-in SMPTE Time Code Reader/Generator, MIDI Time Code synchronization and a digital mixer with pan and level controls. A coaxial S/PDIF digital I/O allows pre-mixed digital bouncing within a single unit, or externally to another recorder or even a DAT or CD recorder. Up to 16 DTRS machines can be synchronized together for simultaneous, sample accurate control of 128 tracks of digital audio.

Features-

- Selectable 16 bit or 24 bit High Resolution audio
- 24 bit A/D and D/A converters
- >104dB Dynamic range
- 20Hz - 20kHz frequency response ± 5dB
- 1 hr. 48 min. recording time on a single 120 tape
- On-Board SMPTE synchronizer - chase or generate timecode
- On-Board support for MIDI Machine Control
- Internal digital mixer with level and pan for internal bouncing, or for quick mixes
- Track slip from -200 to +7200 samples
- Expandable up to 128 tracks (16 machines)
- Word Sync In/Out/Thru
- Analog output on DB25 balanced or RCA unbalanced
- Digital output on TDIF or 2 channels of S/PDIF

CD RECORDERS & DUPLICATORS

MICROBOARDS StartREC Digital Audio Editing/ CD Duplication System

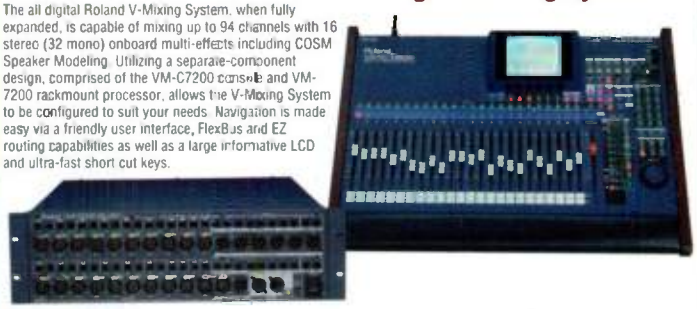
The Microboards StartREC is the first digital audio editing system combined with a multidrive CD recordable duplication system for professionals. Audio is recorded to the internal 6.2 GB IDE hard drive using analog or digital inputs. Sample rate conversion is automatic. Tracks can be edited and sequenced using the StartREC's user friendly interface and up to 4 CDs can be recorded simultaneously. StartREC is the ideal solution for studio recording, mastering, post production or any pro audio environment requiring digital audio editing and short run CD-R duplication.

Features–

- 2X, 4X, or 8X recording speeds
- 6.2GB IDE hard drive
- Editing functions include move, divide, combine or delete audio tracks, add or drop any index or sub index, and create track fade in or fade out
- Coaxial SP/DIF or AES/EBU digital input plus optical S/PDIF I/O
- XLR balanced and RCA Line inputs and outputs
- Automatic sample rate conversion from 32 and 48kHz
- Automatic CD Format Detection feature and user friendly interface provide one touch button operation
- Front panel trim pot and LCD display provide accurate input signal and time lapse metering
- SCMS (Serial Copy Management System) is supported, regardless of the source disc copy protection status
- **StartREC Models Include:** ST2000 (2) 8x writers, ST3000 (3) 8x writers and ST4000 (4) 8x writers

DIGITAL MIXERS

Roland VM Basic 72 Digital Mixing System



The all digital Roland V-Mixing System, when fully expanded, is capable of mixing up to 94 channels with 16 stereo (32 mono) onboard multi-effects including COSM Speaker Modeling. Utilizing a separate-component design, comprised of the VM-C7200 console and VM-7200 rackmount processor, allows the V-Mixing System to be configured to suit your needs. Navigation is made easy via a friendly user interface, FlexBus and EZ routing capabilities as well as a large informative LCD and ultra-fast short cut keys.

Features-

- 94 channels of digital automated mixing (fully expanded)
- Up to 48 channels of ADAT/Tascam T-DIF digital audio I/O with optional expansion boards and interfaces
- Separate console/processor design
- Quiet motorized faders, transport controls, total recall of all parameters including input gain, onboard mixer dynamic automation and scene memory
- 24 faker groups, dual-channel delays, 4-band parametric channel EQ + channel HPF
- FlexBus and "virtual patchbay" for unparalleled routing flexibility
- Up to 16 stereo (or 32 mono) multi-effects processors using optional VS8F-2 Effects Expansion Boards (2 stereo effects processors standard)
- COSM Speaker Modeling and Mic Simulation technology
- 5.1 Surround mixing capabilities
- EZ Routing allows mixer settings to be saved as templates
- Realtime Spectrum Analyzer checks room acoustics in conjunction with noise generator and oscillator
- Digital cables between processor and mixer can be up to 100 meters long- ideal for live sound reinforcement

Options-

- **VS8F-2 Effects Expansion Board** -- Provides 2 stereo effects processors including COSM Speaker Modeling. Up to 3 additional boards can be user-installed into the VM-7200 processor, for 8 stereo or 16 mono effects per processor.
- **VM-24E I/O Expansion Board** -- Offers 3 R-Bus I/Os on a single board. Each R-Bus I/O provides 8-in/8-out 24-bit digital I/O, totalling 24 I/O per expansion board.
- **DIF-AT Interface Box for ADAT/Tascam** -- Converts signals between R-Bus (VM-24E expansion board required) and ADAT/Tascam T-DIF. Handles 8-in/8-out digital audio. 1/3 rackmount size.
- **VM-24C Cascade Kit** -- Connects two VM-Series processor units. Using two VM-7200 processors cascaded and fully expanded with R-Bus I/O, 94 channels of audio processing are available.

EFFECTS PROCESSING

Lexicon MPX-500 24-Bit Dual Channel Effects Processor

The MPX 500 is a true stereo 24-bit dual-channel processor and like the MPX100 is powered by Lexicon's proprietary Lexichip and offers dual-channel processing. However, the MPX 500 offers even greater control over effects parameters, has digital inputs and outputs as well as a large graphics display.

- 240 presets with classic, true stereo reverb programs as well as Tremolo, Rotary, Chorus, Flange, Pitch, Detune, 5.5 second Delay and Echo
- Balanced analog and S/PDIF digital I/O
- 4 dedicated front panel knobs allow adjustment of effect parameters. Easy Learn mode allows MIDI patching of front panel controls.
- Tempo-controlled delays lock to Tap or MIDI clock

t.c. electronic M-One Dual Effects Processor

The M-One allows two reverbs or other effects to be run simultaneously, without compromising sound quality. The intuitive yet sophisticated interface gives you instant control of all vital parameters and allows you to create awesome effects programs quickly and easily.

- 20 incredible TC effects including, Reverb, Chorus, Tremolo, Pitch, Delay and Dynamics
- Analog-style user interface
- 100 Factory/100 User presets
- Dual-Engine design
- 24 bit A/D-D/A converters
- S/PDIF digital I/O, 44.1-48kHz
- Balanced 1/4" Jacks - Dual I/O
- 24 bit internal processing

D-Two Multitap Rhythm Delay

Based on the Classic TC2290 Delay, the D-Two is the first unit that allows rhythm patterns to be tapped in directly or quantized to a specific tempo and subdivision.

- Multitap Rhythm Delay
- Absolute Repeat Control
- Up to 10 seconds of Delay
- 50 Factory/100 User presets
- 24 bit A/D-D/A converters
- S/PDIF digital I/O, 44.1-48kHz
- Balanced 1/4" Jacks - Dual I/O
- 24 bit internal processing

MICROPHONES

B&H is proud to announce that we are now dealers for RØDE microphones and all other Event products.

RØDE NT-2 Condenser Mic

The RØDE NT2 is a large diaphragm true condenser studio mic that features both cardioid and omnidirectional polar patterns. The NT-2 offers superb sonic detail with a vintage flavor for vocal and instrument miking. Like all RØDE mics the NT-2 is hand-assembled in Australia and is available at a breakthrough price.

Features

- Dual pressure gradient transducer
- Large diaphragm (1") capsule with gold-sputtered membranes
- Low noise, transformerless circuitry
- Omni and cardioid polar patterns
- High pass filter switch
- -10dB pad switch
- 20Hz-20kHz frequency response
- 135dB Max SPL
- Gold plated output connector
- Gold plated internal head pins
- Shockmount, Flight Case, and Pop Filter included

SHURE KSM-32SL Cardioid Condenser Mic

The reviews are raving about Shure's new "classic" microphone. The KSM32 features Class A, transformerless preamplifier circuitry, low self-noise and increased dynamic range, all necessary for critical studio recording. It has a 15 dB attenuation switch for handling high SPLs, making it suitable for a variety of sound sources including vocals, acoustic instruments, ensembles and overhead miking of drums and percussion. For studios, the **KSM32/SL** has a light champagne finish and includes an aluminum carrying case, shock and swivel mounts and a velvet pouch. For live applications, the **KSM32/CG** has a charcoal grey finish and includes a swivel mount and padded zipper bag.

- **Frequency response 20Hz - 20kHz**

AKG C4000B Electret Condenser Mic

This new mic from AKG is a multi polar pattern condenser microphone using a unique electret dual large diaphragm transducer. It is based on the AKG Solid Tube design, except that the tube has been replaced by a transistorized impedance converter preamp. The transformerless output stage offers the C4000B exceptional low frequency response.

FEATURES-

- Electret Dual Large Diaphragm Transducer (1st of its kind)
- Cardioid, hypercardioid & omnidirectional polar patterns
- High Sensitivity
- Extremely low self-noise
- Bass cut filter & Pad switches
- Requires 12, 24 or 48 V phantom power
- Includes H-100 shockmount and wind/pop screen
- **Frequency response 20Hz to 20kHz**

GT Electronics AM-61 Cardioid Tube

The GT Electronics AM61 offers classic tube performance in a fixed cardioid, large diaphragm condenser mic. An outstanding addition to any project studio or large commercial recording facility seeking rich, warm tube sounds and unsurpassed value.

- Groove Tubes military-spec GT5840M vacuum tube preamplifier
- Large-diameter, super-thin 3 micron gold evaporated Mylar diaphragm
- Fixed cardioid polar pattern response
- Switchable -10dB attenuation pad and 80Hz low frequency roll-off filter
- Includes hard-shell case, shock mount, hard mount, 6-pin cable and external power supply
- **Frequency response 20Hz - 20kHz**

• **ALSO AVAILABLE** AM-62 multipattern tube condenser mic

audio-technica. AT4047SV Cardioid Condenser Mic

The AT4047 is the latest 40 Series large diaphragm condenser mic from Audio Technica. It has the low self noise, wide dynamic range and high sound pressure level capacity demanded by recording studios and sound reinforcement professionals.

- Side address cardioid condenser microphone for professional recording and critical applications in broadcast and live sound
- Low self noise, wide dynamic range and high SPL
- Switchable 80Hz Hi Pass Filter and 10dB pad
- **Includes** AT8449/SV shockmount
- **Also includes a limited edition tweed flight case while supplies last!**

STUDIO MONITORS

VERGENCE A-20 Studio Reference Monitor System

Incorporating a pair of 2-way, acoustic suspension monitors and external, system-specific 250 watt per side control amplifier, the A-20 provides a precise, neutral studio reference monitoring system for project, commercial and post production studios. The A-20's control amplifier adapts to any production environment by offering control over monitoring depth (from near to far field), wall proximity and even input sensitivity while the speakers magnetic shielding allows seamless integration into today's computer based studios.

- Type Modular, self-powered near/mid/far-field monitor
- 48Hz - 20kHz frequency response @ 1M
- Peak Acoustic Output 117dB SPL (100ms pink noise at 1M)
- XLR outputs from power amp to speakers
- Matched impedance output cables included.
- -6dB LF Cutoff 40Hz
- 5 position wall proximity control
- 5 position listening proximity control between near, mid and far-field monitoring
- Power, Overload, SPL Output, Line VAC and Output device temperature display

Amplifier

- Amplifier Power 250W (continuous rms/ch), 400W (100ms peak)
- XLR, TRS input connectors
- Headphone output
- 5-position input sensitivity switch with settings

Speakers

- 2-way acoustic suspension with a 6.5-inch treated paper woofer and a 1-inch aluminum dome tweeter
- Fully magnetically Shielded with an 18-inch recommended working distance

EVENT ELECTRONICS PS-5 Bi-Amplified Project Studio Monitors

The PS-5s are small format, full-range, non-fatiguing project studio monitors that give you the same precise, accurate sound as the highly acclaimed 20/20 series studio monitors. The use of custom driver components, complimentary crossover and bi-amplified power design provides a wide dynamic range with excellent transient response and low intermodulation distortion.

FEATURES-

- 5-1/4-inch magnetically shielded mineral-filled polypropylene cone with 1-inch diameter high-temperature voice coil and damped rubber surround LF Driver
- Magnetically shielded 25mm diameter ferrofluid-cooled natural silk dome neodymium HF Driver
- 70 watt continuous LF and 30 watt continuous HF amplification per side
- XLR-balanced and 1/4-inch (balanced or unbalanced) inputs
- 52Hz-19kHz frequency response ±3dB
- 2.6kHz, active second order crossover
- Built in RF interference, output current limiting, over temperature, turn-on transient, subsonic filter, internal fuse protection
- Combination Power On/Clip LED indicator
- 5/8" vinyl-laminated MDF cabinet

KRK V-6 Bi-Amplified Near Field Studio Monitors

These bi-amped studio monitors from KRK supply 90 watts of clean power. Their 6-inch woofer & 1-inch silk dome tweeter ensure consistency from top to bottom with crystal clear highs and a solid bass response.

FEATURES-

- 58Hz - 22kHz frequency response
- 1-inch silk dome tweeter and 6-inch long stroke, polyvinyl woofer
- 30 Watt HF & 60 Watt LF amplification
- Magnetically shielded
- Variable system gain +6dB -30dB
- Neutrik XLR/1/4" TRS combo connector

Also Available- V-8

- 1-inch Silk Dome tweeter and 8-inch Woven Kevlar woofer
- 47Hz - 23kHz frequency response
- 60 Watt high frequency and 120 Watt low frequency amplification
- HF adjust -1dB, Flat, -1dB
- LF adjust -3dB at 45, 50 and 65 Hz

Hafler TRM-6 Bi-Amplified Near Field Studio Monitors

Offering honest, consistent sound from top to bottom, the TRM-6 bi-amplified studio monitors are the ideal reference monitors for any recording environment whether tracking, mixing or mastering. Supported by Hafler's legendary amplifier technology that provides a wide and accurate sound field, in width, height and also depth.

FEATURES-

- 33 Watt HF & 50 Watt LF amplification
- 1-inch soft dome tweeter and 6.5-inch polypropylene woofer
- 55Hz - 21kHz Response
- Magnetically Shielded
- Electronically and Acoustically Matched

Also Available- TRM-8

- 1-inch soft dome tweeter and 8-inch polypropylene woofer
- 45Hz - 21kHz frequency response ±2dB
- 75 Watt HF, 150 Watt LF amplification

ALL ITEMS ARE COMPLETE WITH ALL ACCESSORIES AS SUPPLIED BY MANUFACTURER

COMPUTER BASED DIGITAL AUDIO SYSTEMS

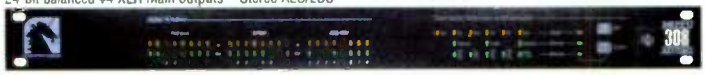
MOTU AUDIO Hard Disk Recording Systems

The MOTU Audio System is a PCI based hard recording solution for the Mac and PC platforms. At the heart of the system is the PCI-324 PCI card that can connect up to three audio interfaces and allows up to 72 channels of simultaneous I/O. Audio interfaces are available with a wide range of I/O configurations including multiple analog I/O with the latest 24-bit A/D/A converters and/or multi channel digital I/O such as ADAT optical and TDIF I/O as well as standard S/PDIF and AES/EBU I/O. Each interface can be purchased separately or with a PCI-324 card allowing you to build a system to suit your needs. Includes drivers for all of today's hottest audio software and AudioDesk, multitrack recording and editing software for the Mac.

THEY ALL FEATURE– • Mac OS and Windows compatible • Includes software drivers for compatibility with all of today's popular audio software plus AudioDesk, MOTU's sample-accurate audio workstation software for Mac OS • Host computer determines the number of tracks that the software can record and play simultaneously, as well as the amount of real-time effects processing it can support • Front panels display metering for all inputs and outputs • AudioDesk Audio Workstation Software for Mac OS features 24-bit recording, multi-channel waveform editing, automated virtual mixing, graphic editing of ramp automation, real-time effects plug-ins with 32-bit floating point processing, crossfades, support for third-party audio plug-ins (in the MOTU Audio System and Adobe Premiere formats), background processing of file-based operations, sample-accurate editing and placement of audio, and more

2408 mkII FEATURES– • 7 banks of 8 channel I/O: 1 bank of analog, 3 banks of ADAT optical, 3 banks of Tascam TDIF, plus stereo S/PDIF. • Custom VLSI chip for amazing I/O capabilities • • Format conversion between ADAT and DA-88 • 8x 24-bit 1/4" balanced analog I/Os • 24-bit internal data bus for full 24-bit recording via digital inputs • Standard S/PDIF I/O for digital plus an additional S/PDIF I/O for the main mix • Sample-accurate synchronization with ADATs and DA88s via an ADAT SYNC IN and RS422

1224 FEATURES– • 24-bit analog audio interface • State-of-the-art 24-bit A/D/A • Simultaneously record and play back 8 channels of balanced (TRS), +4 dB audio • 24-bit balanced +4 XLR main outputs • Stereo AES/EBU digital I/O • Word clock in/out • Dynamic range of 116 dB (A-weighted) • Front panel displays six-segment metering for all inputs and outputs • Headphone jack with volume knob



308 Features– • 8 channels of coaxial S/PDIF using 4 RCA input and 4 RCA output connectors • 8 channels of optical S/PDIF using 4 toslink input and 4 toslink output connectors • 8 channels of AES/EBU using 4 XLR male and 4 XLR female connectors • Word Clock I/O allows the 308 to synchronize with digital audio environments

24i Features– • 24 high quality, 24-bit analog inputs • Balanced 1/4" analog outputs • Optical and coaxial S/PDIF outputs • Front panel headphone output with level control • Word Clock I/O • Connect up to three 24i rack I/Os to a PCI-324 audio card for a total of 72 inputs and six outs

digidesign / AVID

A Division of Avid Technology, Inc

DIGI001 Digital Audio Workstation For Mac And PC

A completely integrated digital recording, mixing and editing environment for the Mac and PC, the DIGI-001 offers a 24-bit multi I/O breakout interface along with Pro Tools LE software— based on Digidesign's world renowned ProTools software. The DIGI-001 interface features 18 simultaneous I/Os made up of 8 analog inputs and outputs— two of the inputs are full featured mic preamps with phantom power, and digital I/O including standard S/PDIF as well as an ADAT optical interface that can also be used as a S/PDIF I/O. ProTools LE supports 24 tracks of 16 or 24-bit audio and 128 MIDI tracks and also features RealTime AudioSuite (RTAS) effects plug-ins. For ease of use, MIDI and audio are editable within the same environment and all mixing parameters including effects processing can be fully automated.

FEATURES–

- 18 simultaneous, 24-bit ins and outs with support for 44.1 and 48 kHz sample rates
- 20Hz - 22kHz freq. response ± 0.5 dB
- 2 channel. XLR mic/1/4" line inputs with -26 dB pad, 48v phantom power, gain knob, and HP Filter at 60Hz
- 6 ch. line inputs (1/4") TRS balanced/ unbalanced w/ software controlled gain
- +4dB balanced 1/4-inch Main outputs
- Balanced 1/4" monitor outs with front panel gain knob
- 1/4-inch unbalanced line outputs channels 3-8
- Headphone output with independent gain control knob
- 2 channel S/PDIF coaxial digital I/O
- 8 channel ADAT optical I/O can also be used as 2 channel optical S/PDIF

Pro Tools LE

- Supports 24 tracks of 16 or 24-bit audio and 128 sequenced MIDI tracks
- Sample-accurate simultaneous editing of audio & MIDI
- Real-time digital mixing capabilities include recall of all mixing parameters, support for edit and mix groups and complete automation of all volume, panning, mutes and plug-ins
- Route and mix outboard gear in realtime
- MP3 and RealAudio G2 file support (Mac)
- Two plug-in platforms offer multiple options for effects processing— Real-Time AudioSuite (RTAS) is a host-based architecture that allows an effect to change and be dynamically automated in realtime as the audio plays back — AudioSuite is a file-based format, that renders a new file with the processed sound.
- Bundled RTAS plug-ins include, 1 and 4-band EQ, Dynamics II- compressor, limiter, gate and expander/gate, Mod Delay - short, slap, medium, and long delays with modulation capabilities for chorus or flange effects and dither. AudioSuite plug-ins include Time Compression/Expansion, Pitch Shift, Normalize, Reverse.

MIDI Functions

- MIDI functions include graphic controller editing, piano roll display, up to 128 MIDI tracks and editing options like quantization, transpose, split notes, change velocity and change duration.
- MIDI data can be edited on the fly

SOFTWARE

Also available with MOTU's award-winning Digital Performer audio sequencer software package

Digital Performer 2.7 MIDI/AUDIO Software for Mac

Digital Performer is an integrated multitrack digital audio and MIDI sequencing program packed with advanced tools for a wide variety of audio applications. Sample accurate editing, loop based audio capture, realtime DSP effects and the best MIDI timing/resolution available insures unlimited creative potential.

FEATURES-

- Includes over 50 real-time MIDI and audio effects plug-ins • POLAR window - which provides Interactive audio loop recording • 24-bit recording and editing • 32-bit native effects processing - incredible sounding EQ and other FX • 64-bit MasterWorks Limiter and Multiband Compressor plug-ins included • Sample-accurate - the most reliable waveform editing and tightest sync you can get • Samplers window - drag & drop samples between your Mac and your Sampler • PureDSP stereo pitch-shifting and time-stretching • Unlimited audio tracks, real-time editing, full automation and remote control • QuickTime digital video support

NEW FEATURES-

- Full Plug-In FX automation and increased 3rd party Plug-in support
- Drum Editor
- **Adjustable Display Resolution** from 2 to 10,000 PPQ. Tick values up to four decimal places can be set allowing 1000 times greater editing resolution. For example, if you are used to editing MIDI data at 480 PPQ, you can set your edit resolution to 480,000 for 1000 times more precision.
- **MIDI Time Stamping** (MTS) which exists in MOTU's rack-mountable USB MIDI interfaces, delivers MIDI data from Digital Performer to MIDI devices as accurately as a third of a millisecond for every single MIDI event.

TC WORKS SPARK 1.5 2-Track Editing For Mac

ULTIMATE SOFTWARE MACHINES

Spark is professional 2-track audio editing software for the Power Macintosh that provides fast access to files and powerful processing tools. Supports files up to 24-bit/96kHz and has batch processing, VST plug-in support, as well as MP3 file export built-in. Audio can be extracted from a Quicktime movie, edited and then exported along with the video to a new file. Bundled with Adaptec's Toast so you can burn your audio directly to CD.

- Browser View- File database, audio editor and play list all in one easy to use display with movable border lines- Eliminates the need for surfing several windows to access and edit files.
- Wave Editor- Perform off-line editing, processing, and create markers and non-destructive regions
- Supports AIFF, Sound Designer, .WAV and QuickTime file formats.
- DSP Processing Includes- Normalize, Reverse, Fades, Crossfades, and Sample Rate conversion and **realtime Time Stretching**
- VST Plug-In compatible
- Supports file swapping with most major samplers and any sampler that supports SMDI • Batch Processing
- Bundled with Adaptec's Toast Pro you can burn your audio on CD
- Extract audio from a quicktime movie for editing and then export the audio along with the video into a new file
- **SPARK 1.5** supports MP3 audio authoring for the web directly from the file menu.

ANTARES AMM-1 Microphone Modeler

The AMM-1 Microphone Modeler uses patented technology to create precise digital models of a wide variety of microphones, from historical classics to modern exotics and even industry-standard workhorses. Simply tell the Microphone Modeler what microphone you are actually using and what microphone you'd like it to sound like. It's as simple as that. Available as a plug-in for the TDM and MAS environments, with DirectX and Mac VST not far behind.

FEATURES–

- Proprietary DSP-based acoustic modeling allows any reasonable quality microphone to sound like any of a wide variety of high-end studio mics • Models reproduce the effects of windscreens, low-cut filters, pattern-dependent frequency response and proximity effects • Create hybrid mics that combine the bass response of one mic with the treble response of another • Add a model of classic tube saturation distortion • Use during mixdown to change the mic on an already recorded track • Incredibly simple to use - simply select the mic you're using and the mic you want it to sound like • Includes an extensive collection of digital models of historical classics, modern exotics, and industry-standard workhorses • Additional models can be downloaded from the Antares web site

WAVES Pro-FX Bundle Plug-ins For Mac or PC

The latest Bundle from Waves has some of the coolest sound design plug-ins available for the Mac and Windows platforms.

SuperTap - Six taps of mono or true stereo delay (up to Six seconds) • Global LFO modulation • 2 feedback modes • Q10-style filtering for each tap • rotation (stereo panning) • Delays are adjustable in milliseconds and note values • Tap out delay times or patterns using the Tap Pad

MetaFlanger - Vintage tape-flanging, phaser-emulation, and special effects • True dual-delay flanging sounds • Wet signal include filters so you can flange or phase just part of the signal • Factory presets of vintage emulations (Mutron, MXR, Itchycoo Park) and more

MondoMod - AM, FM, and Rotation (stereo panning) modulators • Gentle wandering guitar solo panning or bizarre destructive effects • Single LFO drives all modulators with independent phase offsets between the modulator signals

UltraPitch - Formant-corrected pitch shifter with 6-voices • Excellent gender-bending • Independent stereo panning and delay • Animator delay-randomizer. Set the pitchshift by musical intervals (with 5 cent resolution) • Manual formant mapping as well as presets that perfectly match instrumental formant responses • Creates huge and thick stereo chorusing, doubling, parallel harmonies, excellent vocal slap/spread effects, and much more

Enigma - Mysterious and innovative, Enigma combines a complex notch filter, short delay feedback loops and modulation to create distinctive sound design effects.

Doppler - Developed for post, film sound and game designers. Provides realtime doppler effects with auto and manual triggering modes, full control of air damping, panning pitch, path curve, gain, start/stop points and reverb tail.

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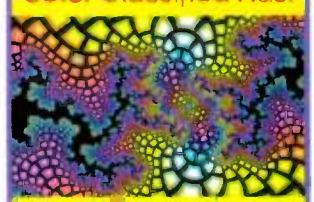
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MOTU AUDIO SYSTEM

AMP SIMULATION

SansAmp™ PSA-1

Co-developed with Tech 21, the SansAmp PSA-1 captures every sonic subtlety of the original. The SansAmp PSA-1 provides the widest range of amplifier, harmonic generation, cabinet simulation and equalization tone shaping options available. It appears on literally thousands of recordings, used on everything from guitar and bass to drums and harmonica! All 49 original SansAmp presets — featuring Marshall®, Mesa Boogie®, Hiwatt®, Fender® Lead, Rhythm and Bass; and Ampeg SVT® sounds — faithfully and authentically recreated. Explore dozens of new presets contributed by well-known artists and producers, then create your own unique tone!

Names of sample settings are intended for descriptive purposes only and should not be construed as an endorsement or affiliation with any companies, products, song titles, or artists named.

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Authentic analog designs from electronic music legend Bob Moog, built with bombfactory™ digital technology! Moogerfoogers combine the best features of audio effects and vintage synth modules, opening new frontiers for sonic exploration. The Lowpass Filter features a 2-pole/4-pole variable resonance filter with envelope follower. Use it to achieve classic '60s and '70s sounds on bass and electric guitar, or dial in warm, fat analog resonance on any instrument. The Ring Modulator provides a wide-range carrier oscillator and dual sine/square waveform LFO. Add motion to rhythm tracks and achieve radical lo-fidelity textures — you set the limits!

CLASSIC COMPRESSION

bombfactory™ Classic Compressors

bombfactory™ Classic Compressors look, sound, and work just like the real thing. Whether you're just learning to use compression or a seasoned pro polishing a final mix, you'll love these time proven designs. Meticulously crafted digital versions of the LA-2A and 1176, the most popular vintage compressors used in top pro studios. Proprietary modeling technology (patents pending) captures every tube, transformer and transistor of the originals! Plus sidechain support and perfect stereo tracking, two features not available in the "vintage" domain.

VINTAGE VOCE FX

Voce™ Spin & Chorus / Vibrato

From Voce, classic effects that sound great on any instrument. Voce Spin provides the most accurate simulation of the well-loved rotating speaker. It's a favorite of producers, guitarists, and, of course, organ players. Fifteen classic recording setups feature different speaker cabinets, varying microphone placement — even the "Memphis" sound with the lower drum's slow motor unplugged! Voce Chorus/Vibrato recreates the B-3 Organ's mechanical scanner vibrato. Three settings of Chorus and three settings of Vibrato on one cool knob. Fun and easy to use, it's a classic effect used for over sixty years. Talk about vintage!

Digital Done Right

THE DESKTOP STUDIO

Digital Performer is the one to own. It's a complete desktop studio environment that gives you: unsurpassed timing accuracy and precision, an entire line of MOTU audio & MIDI interfaces to support it, a wide range of third-party plug-ins and hardware peripherals designed especially for Digital Peformer, and the most complete set of advanced features currently available to track, edit, mix, process and master your MIDI and audio recordings. Track your mixes through our new 2408mkII audio interface — now with balanced quarter-inch, 24-bit analog I/O (8 in / 8 out), with inputs that are switchable between +4 /-10, plus a volume knob for the main outs. Same price. Same incredible product. Just more value.

MOTU

STEREO REVERB

RealVerb™

On the heels of their ground-breaking RealVerb 5.1™ surround reverb plug-in, Kind of Loud Technologies presents RealVerb™, a new stereo reverb plug-in for MAS. RealVerb uses complex spatial and spectral reverberation technology to accurately model an acoustic space. The bottom line? Great sounding reverb with the ability to customize a virtual room and pan within the stereo spectrum. RealVerb even lets you blend room shape, material, and size according to the demands of your mix. And RealVerb was designed from the ground up for automation: adjust controls in real-time without distortion, pops, clicks or zipper noise. You can even morph between presets – in real-time. Don't rely on your old standby – let RealVerb bring new quality and space to your recordings.

KIND OF LOUD

ESSENTIAL PROCESSING

Waves Gold Native Bundle™

It's everything you need, with essential daily tools, sweetening and mastering processors, and sound design mindbenders. From the original Q10 and L1, to the Renaissance series, to Enigma and MondoMod. Don't skimp. Go for the Waves Gold, on the way to your gold record.

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- Native Power Pack
- Native Power Pack II
- Pro-FX Plus

...a total of 15 Waves processors... and you save a bundle when you go for the gold!

WAVES

24-BIT SAMPLING

Unity DS-1™

Unity DS-1 is software that turns your computer into a full-featured, professional digital sampler. With Unity DS-1, you can recreate the sounds of acoustic instruments or any other audio source with stunning realism and control. Unity DS-1 was designed for musicians by musicians. We built a real stereo sampler with the ability to load huge samples in seconds instead of minutes. We also made sure it had lightning fast note-on response time. Unity DS-1 can re-create the sounds of acoustic instruments or any other audio source with an extensive MIDI implementation for real-time control of all parameters and the best integration with Digital Performer in the industry. So whether you need a multi-timbral sound module at home, or a live performance 24-bit sampler for the road, Unity DS-1 brings it all together.

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MOTU AUDIO SYSTEM

CONSOLE PROCESSING

ChannelStrip

The Ultimate Integrated Bundle

The word is out...

"I am happy to report that ChannelStrip is working very smoothly with Digital Performer 2.7. The first thing I noticed was how clear and exacting the EQ sounds, nothing else adds the "air" this plug has. Add to this the side-chainable gate and compressor and you'll find that nothing else offers so much efficiency from a single insert slot. The CS plug is also very MAS friendly, exhibiting consistent and reliable performance. Now that we have plug-in automation, I will be exploring dynamic EQ moves as opposed to multing the audio to multiple channels on my console. I often add some highs to my lead vocals when I hit a chorus so they will cut through better, now all it takes is an automation move, pretty cool. There are many applications for this plug, I will share some in the future and encourage others to do so as it will only strengthen our collective experience with DP." —Steve Meyer

"I LOVE the MIXES I am getting with da ChannelStrip. It really makes mixing a pleasure. Hey – I got a $250,000 Euphonix just sitting there looking pretty; NOW WHAT !?!?!" —Giorgio Bertuccelli

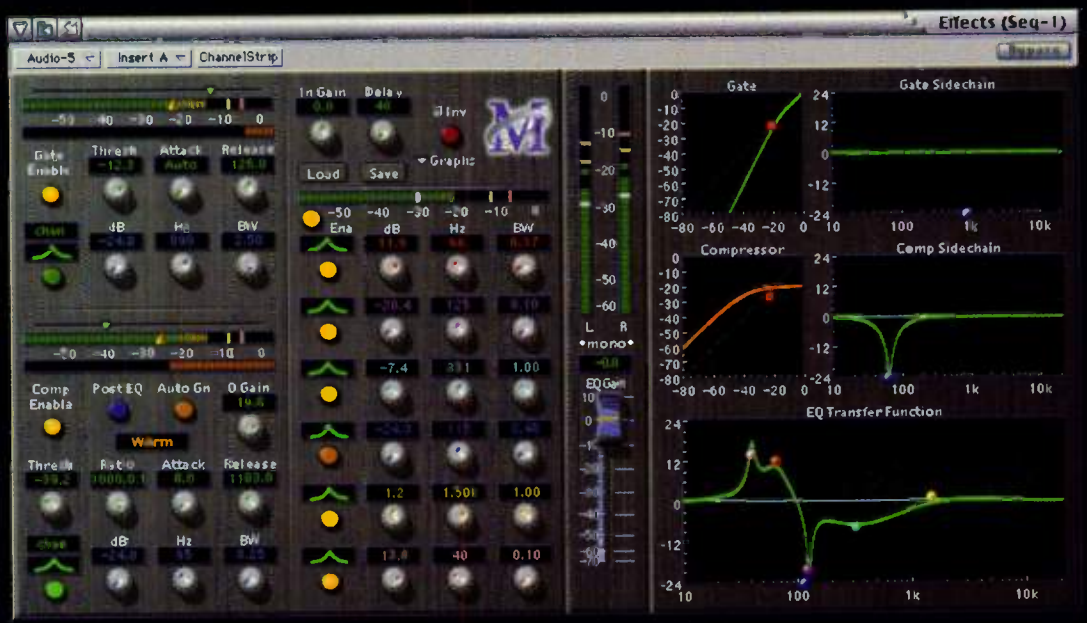
"Wow, it really sounds great! Finally, real EQ in software. I love this thing!" —Jim Watson, FAT GROOVE Productions

"Sonically, I'm knocked out with ChannelStrip — I am really really critical of a lot of the plug-in stuff and work with some very demanding artists at some of the finest studios around, plus my own room which is about as good as it gets for overdubs and mixing. I would not think twice about using ChannelStrip on anything in front of anyone — in my limited time with it, I think it's that good. You're onto a gold mine with this. I've loaded up a few tracks only so far and used the automation to do a few things I normally have to do in real time. Unreal... Once again, it's the sound of ChannelStrip that I can't get over. The controls, layout, etc. are very cool, too." —Jack Hale

"The whole plug-in gives the impression you looked at an SSL pretty close. I've always been an SSL-man, but not any longer! Still can't believe my luck.... Nice work!" —Steve Rhodes, RME

ChannelStrip is available NOW for Digital Performer, MOTU 2408 and all MAS 2.1 compatible DAW's.

Don't delay! Get one of the hottest plug-ins for MAS today.



METRIC HALO LABORATORIES

SCALABLE REVERB

EverPack Bundle with Z•Room Reverb™

From tube warmth to stereo enhancement, there's no other way to get the smooth sound produced by these classic MAS plug-ins from DUY. And now DUY has added an entirely new piece to the EverPack bundle: Z•Room. This plug-in gives you an entirely new approach to stereo reverb, with 64-bit processing that produces high-quality density and diffusion for unparalleled sound quality, total control over parameters never before available, 'deluxe mode' for maximum sound quality, 'economy mode' for maximum efficiency, 'earlies' mode for enhancing the effect of early reflections, and the exclusive 'Rehearsal Mode', which lets you to set parameters while listening to the real impulse response of the reverb— an unprecedented tool for evaluating reverb. For more info: www.duy.com.

DUY

SESSION BACKUP

Mezzo™

You've got a Digital Performer system that produces gigabytes of crystal-clear, digital audio and sequence data. You know that regularly backing it all up is important, but how can you get it organized and archived without wasting studio time? That's what Mezzo is all about – automated, project-based backup of your data. A DP project can contain hundreds of separate audio files – generic backup programs can't track these files on a per-project basis, but Mezzo can. And with full background operation you can backup or restore while you compose! With its intuitive, drag & drop interface and practically hands-free operation, Mezzo makes the job of managing the daily flow of data a simple and painless task.

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FINAL MIX

By Larry the O

Stale Beer and Bitter Fruit Make Poor Nutrition

All of us experience defeat in our lives, and those of us working in music and sound experience it all too often. We can all recall the lost gig, the band that fell apart just as it started to break through, and our own mondo goof-up on an important mix. Maybe you were left feeling foolish or holding the bag. "In any event," you think, "I won't let *that* happen again."

Well, maybe it won't happen again, but the method of achieving that extra bit of protection often turns out to be raising a barrier that puts a safe distance between you and other people you work (or live) with. Even when the potential cost of not protecting oneself far outweighs the slight chill the barrier brings, it's still another brick in the wall.

By the time you've been roughed up a few times in business, you start to get tough and expend no small amount of energy guarding against potential vulnerabilities. Behind the scenes, the phantom menace creeps into your psyche as the folly of it all grows old and a cynical streak appears and widens with each bad experience, until pessimism and cynicism dominate your view. Once that happens, the thrill is gone, and you're headed for the exit. Your eye rolling and constant whining are unpleasant to be around and result in self-fulfilling prophecies of doom.

The situation sounds pretty desperate, but most of us struggle with this to a greater or lesser degree. Let's face it: the business of music and sound is brutal, and the rewards, while potentially great, are few and far between in comparison to the overwhelming difficulties and endless work required to turn a creative idea into a finished work or performance. Is there an alternative to this bleak scenario? If so, what trade-offs are entailed?

In the realm of human consciousness, there are no magic bullets or perfect solutions, but there can be hope. The most effective tactic I've found (not to suggest I have conquered burnout and bitterness) feels like a martial-arts technique. Essentially, I try to find a way to let negativity and anger pass through me and be carried away on the winds of time.

The first step is simply accepting the situation: something rotten happened, I'm frustrated and angry about it, and I'd like to have my revenge, get what I earned, or otherwise recover what I feel was taken from me. Refusing to let that be the case only causes me to internalize those feelings, which later re-emerge in a hundred destructive ways.

Once I've accepted events and my feelings about them, I am poised to change. If there is any rectification for some or all of the situation (revenge is *not* rectification), this is the time for me to identify it and do what I can. If nothing can be done, I can only move on.

The next—and, for me, the biggest—step is to find The Lesson. Every experience offers a lesson. Not all lessons are positive statements, but a lesson learned suggests a point to the whole exercise. This fixes nothing, but it does enable me to release a lot of "Why me?" feelings.

Finally, I try to put the lesson to use to help me see similar situations coming long before they happen and, when possible, simply sidestep them. When I can't get out of the way, then I may need to put up the barrier, but hopefully with an understanding full enough to let it be purposeful and temporary.

When these things work, I carry much less baggage in my career and life. When they don't, I feel like old Jacob Marley's ghost, adding another link to the chains I forge in life.

Larry the O *provides music and audio services with his company, Toys in the Attic, and is a sound designer at LucasArts Entertainment.*

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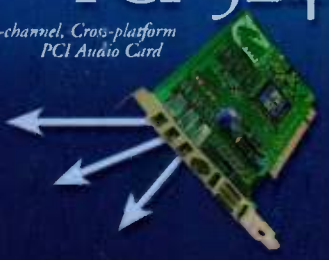

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