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# Electronic Musician

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May 2000

PRO RECORDING TIPS FOR YOUR NEXT GIG

## **SPDIFAESEBUADATTDIE**

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100 Years of Electronic Music,
Part 4: The Future

Roland VM-3100Pro and 13 other reviews

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

-Roger Nichols, EQ Magazine

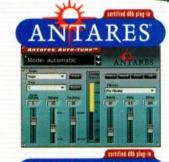
PLUS 3 MORE REASONS TO GO FOR IT.

## 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.

- New key (sidechain) inputs for all 48 onboard dynamic processors featuring soft knee architecture and single band 20-20k parametric EQ for frequency dependent processing such as de-essing
- 3rd-party plug-ins via our new UFX card. Up to 16 simultaneous plug-ins on the first 48 channels, pre or post DSP, pre-fader via up to 4 UFX cards. Each plug-in is available twice once when tracking, and again at mixdown!
- Multiple Undo List 999 levels!
- New Snapshot libraries.
- 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.
- Cross patching allows substitution of channels between various banks.

The list of top engineers and producers who use the awardwinning Mackie Digital 8 . Bus is growing daily. For info on the D8B, new UFX and Optical • 8 cards, 3rd-party plug-ins and how D8B owners can get their free OS upgrade, visit www.mackie.com or call your local D8B dealer.













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 OIV. Thanks to all who helped us win this prestigious award.



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-touse 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.

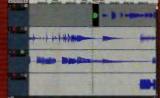
TC Electronic Reverb (bundled with the D8B UFX card) provides Reverb I and Reverb 2 algorithms from the renowned TC Electronic M2000 Studio Effects Processor. TC FX upgrade package contains an expanded set of M2000 reverbs plus Delay, Chorus, and Pitch. TC 2000 adds the TC M2000's Reverb 3, de-essing,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





www.mackie.com - 800/258-6883

## HDR24/96. MACKIE'S NEW 24 TRACK RECORDER. **WORKS WITH ANY MIXER. NO EXTRA** COMPUTER OR SOFTWARE NEEDED.







HDR24/96 editing features include:

8 takes per track with nondestructive comping, nondestructive cut/copy/ paste of tracks, regions or super-regions, drag-and-drop fades & crossfades, 1x/2x/4x/8x/24x wave-



form views, bidirectional cursor scrub and unlimited locators and loops... with unlimited undos - but without requiring an external computer! Coming soon: DSP time compression/

expansion, true waveform editing with pencil tool, invert, pitch shift, normalize and much, much more.

- Built-in 20-gig Ultra-DMA hard disk plus front panel bay for additional easily available pullout drives
- · Intuitive analog tape deck interface and monitoring
- · Syncs to SMPTE, MIDI, Black Burst, PAL & NTSC without extra cards
- Unlimited HDR24/96 linking! Sync 48, 72, 96, 128 or more tracks sample accurately
- 96kHz recording via software and new PDI 96 I/O
- Digital 8 Bus I/O cards mix and match!
- 3.5-inch disk drive for software upgrades & tempo map importing
- · Fast Ethernet port built-in
- · Remotes available.

- 24 tracks...24-bits
- · Built-in full-feature digital workstation editing
- · Affordable pull-out media
- · Built-in SVGA, mouse & keyboard ports
- Built-in IOOBaseT Ethernet

ew hard disk recorders are popping up all over the place.

Our new HDR24/96 is the only recorder with built-in nondestructive graphic waveform editing. Just plug in a mouse, keyboard and SVGA monitor to view all recorder parameters on screen in real time. Enjoy complete editing control with unlimited levels of undo, drag-and-drop crossfades with 9 preset combinations plus fade/crossfade editor. And look forward to DSP time compression/ expansion, pitch shift and lots more!

The HDR24/96 was the only recorder that uses pull out Ultra-DMA hard drives, so affordable that you can keep one for each project—over 90 minutes of 24-track recording time costs less than a reel of 2-inch tape!

Call or visit our website for preliminary info on the new HDR24/96. Shipping soon from Mackie Digital Systems.





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## Step into the cakewalk

## Pro Sute

Cakewalk Pro Suite is the complete Windows workstation for multitrack recording, real-time mixing, and hard-disk based sampling. Once inside, you're free to produce professional music and sound projects entirely in the digital domain.

The Pro Suite provides an arsenal of software tools for today's recording musician. It combines essential recording and sampling software technologies into an integrated studio solution. There's nothing else like it available today.

## CAKEWALK PRO AUDIO" 9

- Record and mix up to 256 tracks of digital audio and MIDI
  - WavePipe low-latency audio mixing and playback using standard Windows audio cards
  - Supports 24-bit/96 kHz audio hardware
  - Exports audio to MP3, RealSystem G2, and Windows Media formats for Internet delivery
- · Notation with guitar tablature, fretboard editing
- · Sync to film and video; import digital video
- · Non-linear, graphical editing of audio and MIDI
- Supports real-time DirectX plug-ins
- Supports real-time MFX MIDI plug-ins
- StudioWare for MIDI-based studio automation

## NEMESYS GIGASAMPLER LE

- · Hard disk-based sampler
- · Integrates with Pro Audio 9 as virtual synthesizer
- Provides gigabyte-size sample sets
- · Loads samples in seconds, not minutes
- Save and load entire performances
- · Fast, tight note-on responsiveness for live playing
- · Sample instruments with full natural decay
- Full looping implementation (although looping is not necessary)
- · 32-bit audio signal processing
- · Reads GigaSampler, .WAV, and Akai" Libraries
- Includes GigaPiano Sample Library CD-ROM



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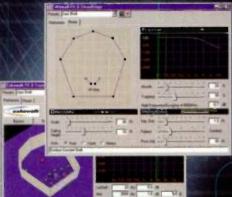
## CAKEWALK AUDIO FX" 1 DYNAMICS PROCESSING

- Compressor/Gate maintains audio signal levels at user-defined levels
- · Expander/Gate increases dynamic range of audio
- Limiter prevents audio signals from exceeding user-defined threshold

### CAKEWALK AUDIO FX" 2 VINTAGE TAPE AND AMP SIMULATION

- Advanced processing algorithms add classic sound and warmth to "dry" and "cold" digital audio tracks
- AmpSim adds guitar amplifier sound to digital audio; choose and modify amp model, speaker cabinet, overdrive, EQ and other parameters
- TapeSim adds tape saturation and natural warmth associated with analog magnetic tape decks





### CAKEWALK AUDIO FX 3 SOUNDSTAGE DESIGN FOR CUSTOM REVERB

- Design virtual rooms in which to process digital audio tracks
- · Add and move walls, adjust ceiling heights, define surface absorption properties
- · Choose microphone types and placements
- · Assign audio tracks to different "performers"
- · Use and modify predefined spaces, like jazz club, arena, and cathedral presets

All Cakewalk Audio FX plug-ins provide:

- · 32-bit, floating point effects processing
- · DirectX-compatibility, the Windows standard
- · Real-time effects processing and off-line editing

## MUSICIAN'S TOOLBOX III

- · 2 CD-ROMs of multimedia content and tools
- · Digital audio loops and samples
- · Professional MIDI drum patterns
- · Digital video images
- · SoundFont instruments
- · Advanced Pro Audio 9 tutorials
- More



Cakewalk Pro Suite is available worldwide. For more information, visit www.cakewalk.com, or call 888-CAKEWALK (617-441-7870 outside U.S.).





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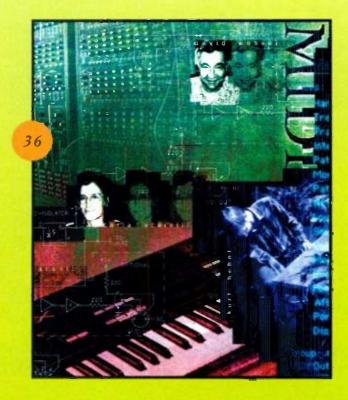
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Special thanks to Bob Coms

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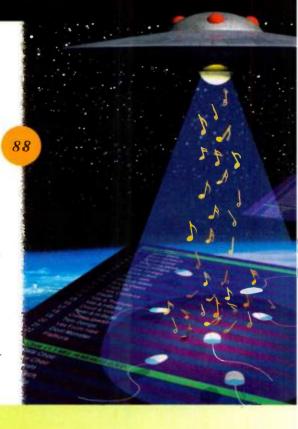
Thanks to the Internet, you'll never run out of fresh sounds again.

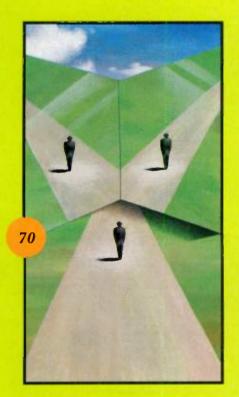
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## **Spring Cleaning**

ere in sunny California, spring comes early. Ready or not, it's time for spring cleaning. For most folks, that means freshening the house, catching up on yard work, and clearing out junk. For musicians, it's also a good time to take a fresh look at our studios—and our music.

Most of us are so busy that we put off full-scale reexaminations of our lives. But we're asking for trouble if we don't occasionally step back and take

a long, hard look at what we're doing, how we're doing it, and how we feel about it. The physical aspects of spring cleaning are obvious. Inventory your music gear, right down to the cables and connectors. Use the list to update your insurance and identify items you want to sell. Check and clean cables, connectors, and patch bays; (carefully) dust the inside of your computer; restock removable media and cleaning materials; and update your wiring charts. (You do have a diagram of all the connections in your studio, right?) Reorganize your removable media, track sheets, song files, and sample library. Review your studio's acoustic treatment and examine its lighting and ventilation. Check everything.

But physical cleaning is just the beginning; spring is also the time to take mental and emotional stock of your studio, your band, and your music. Make a list of what you like and don't like about your musical life, keeping in mind that life involves change and growth, so the tasks you enjoy now might not be the same ones you liked in the past.

For instance, are you happiest when engineering, producing, writing, arranging, designing sounds, performing live, playing sessions, or doing business? Perhaps you've spent so much time working with clients that you haven't written a new tune in months. Maybe you've let your studio become an ergonomic mess. And don't overlook music-business issues and your band's structure and musical direction. You probably can't change everything you dislike, but moving closer to your ideal will help keep you happy and productive.

As you examine your musical life, ask yourself one question: do I still carepassionately, as a creative artist should—about my work? If not, why? For instance, it's okay to craft music strictly for the money, but are you neglecting your artistic side? Perhaps your musical tastes are changing, and you need a fresh challenge. Don't wait until the thrill is gone before you think about it.

Some people avoid self-examination until they wake up one day and realize they're miserable. I hope you won't do that. Spring is a time of rebirth and renewal, so seize the opportunity to reinvigorate your musical life.

•••••

Speaking of reinvigoration, EM is getting a fresh charge of positive energy from new assistant editor Marty Cutler, who has more or less inherited Rick Weldon's old gig, including the "What's New" column. (Weldon now writes for Dolby Labs.) A master of and innovator on the 5-string banjo, Cutler is a computer-savvy veteran electronic musician and recording artist. He plays a wide variety of music, created MIDI files for PG Music's The Bluegrass Band, and is an authority on MIDI guitar and—yes, seriously—MIDI banjo. Cutler is also a strong writer who fits in perfectly at EM.

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# roll your own.



Ever wish that you could take your studio on the road? Keeping a piece of gear around for just one sound? E-MU now lets you "roll your own" custom sound ROMs for the Proteus® 2000 using any E-MU® Ultra Sampler. Fill the Proteus 2000 with custom sound ROMs, and your dream of instant access to all your favorite sounds in a portable, permanent package is a reality!

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- 1. Load or sample your favorite sounds into your E-MU Ultra Sampler
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Featuring 128 voice polyphony, expansion up to 128 MB of internal sounds, 32 MIDI channels, and a lightning-fast processor, Proteus 2000 is already the platform of choice for demanding musicians around the world. Whether you expand your universe of sounds by "rolling your own" custom ROMs or take advantage of E-MU's expansive library of "pre-rolled" sound ROMs, the Proteus 2000 will deliver superior performance at a price you can afford.

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## ORCHESTRAL SESSIONS NEW VOL. 1: SECTION STRINGS \$395 MSRP

The 32 MB Orchestral Sessions Vol. 1 expansion ROM offers you the most realistic and comprehensive collection of section strings available. Meticulously sampled and intutively arranged, these new string samples include various bowing styles, pizzicato, tremolo and other effects for all section strings.



## ORCHESTRAL

SESSIONS NEW VOL. 2: WOODWIND, BRASS, PERCUSSION AND SOLO STRINGS \$395 MSRP

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The 32 MB Sounds of the ZR expansion ROM faithfully reproduces the diverse sounds of ENSONIO's popular ZR-76 keyboard, including William Coakley's Perfect Piano. From synths and drumkits to one of the finest pianos ever sampled, this collection delivers ENSONIO's finest sounds at a tremendous value.

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1600 Green Hills Rd., Scotts Valley, CA 95067 Tel: 831.438.1821 Fax: 831.438.8612 www.emu.com ince we introduced the MX-2424 hard disk recorder, there has been a lot of speculation about its price (which is so low it seems too good to be true).

So we get questions. Like...

"24 tracks is an upgrade?" (No, it's 24 tracks right out of the box.)

"24-bits is an upgrade?" (No, all the bits are there too.)

"Do I have to pay extra for inputs and outputs?" (No. At \$3,999 estimated street price\* you get a full set of 24 TDIF-1 or ADAT\* optical digital inputs and outputs — plus an assignable stereo AES/ EBU - S/PDIF pair. For a little more you can get 24 channels of AES/EBU digital I/O, or analog — or both digital and analog!)

"Does it need an external computer?" (No. The MX-2424s front panel has a full set of professional

transport, editing, and track assignment controls, including a shuttle/ scrub knob. So you don't have to have a computer to run it. But — if you happen to own a Mac or a PC, you can take advantage of the digital audio editing and control software that comes standard with each MX-2424 to do even more. Your choice.)

"Before I start recording do I need to buy a monitor, a keyboard, or a hard drive? Or anything else?" (No. Nyet. Nope. Not at all. Just hook up power and start recording.)

So let's make this as plain as we can: The MX-2424 is an amazing, full-featured professional 24-track digital recorder. And there's never been anything like it at this size or price.

Its sonic performance is outstanding. Lots of companies claim 24-bit 48k performance, but only the MX-2424 is part of TASCAM's M Series family of multitracks — the products chosen for their sonic performance by such discriminating facilities as Skywalker Sound, Universal Studios, and 20th Century Fox.



\$3,999

Superior reliability is guaranteed. The MX-2424 was designed from the bottom up to be a great recorder, and nothing but a great recorder. Its processors and circuitry are fully optimized for audio - not video games, spreadsheet software, or surfing the web. And isn't that absolute focus and rock solid performance exactly what your music deserves? Over the last three decades we've designed and built literally millions of professional recorders and recording systems; the MX-2424 is the culmination of everything we've learned.

Of course that way you'd miss the great light show from the 24 tracks of level metering and channel status displays... but the real point here is simplicity. When you want the MX-2424 to start recording, just reach over and press REC + PLAY (just like a traditional tape recorder). In a fast-paced production environment, you can record to hard drives that mount into standard Kingston® carriers and plug into the front panel drive bay. Just pop in a new drive at the start of each session. It doesn't get any simpler than that.

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# Really.

The power to meet your needs. A standalone MX-2424 is an incredibly powerful unit, with enough internal hard disk capacity to hold about 45 minutes of 24-bit 24-track audio. The MX-2424's Fast/Wide SCSI port lets you connect up to 15 external drives and record directly to all of them. And if you need more than 24 simultaneous tracks, just add additional MX-2424's. Up to 32 MX-2424's can be locked together in sample accurate sync to act as a single recorder.

Professional recorders need to interface with increasingly complex systems.

✓ It provides video and time code lock capabilities as standard features, making it easy to integrate with external workstations.

✓ It resolves to AES/EBU, S/PDIF, word clock, TDIF-1, ADAT optical, SMPTE Time Code (LTC), and video, and chases MIDI Time Code.

✓ Available Input/Output modules include TDIF-1, AES/EBU, ADAT optical, and analog. It's a complete professional hard disk multitrack in a portable, affordable, rackmount box. You can plug it in, turn it on, and start recording.

✓ Back panel ports include Fast/Wide SCSI, ethernet, MIDI, RC-2424 remote, and TI-BUS!

Want a remote control? Get the one that's made to take advantage of the power in your MX-2424. The RC-2424 remote is a powerful, professional multi-machine controller with all of the MX-2424's front panel features, plus macros and more.

MX-2424 shipments are about to start, and there is already a waiting list. To get yours sooner instead of later, contact your authorized TASCAM dealer!

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## **MP3 MISUNDERSTANDING**

got into an argument with my school's computer administrator about MP3s and CD burning. He told me that, with Adaptec's Easy CD Creator. you can use MP3 compression to write an audio CD, and that, due to the MP3 compression, the audio playback will take up about one-tenth the space it occupied when uncompressed (just as MP3 files use about one-tenth the storage space of CD-quality WAV files). He also claims that the disc will be playable in a conventional CD player. You can therefore fit the contents of ten audio CDs on a single disc by ripping everything to MP3 and then writing it to the disc.

I tried to tell him that the MP3s must be reconverted to CD-audio format when they are written to the disc to allow CD-audio playback. I think that if I burn MP3s to CD, I need to play them back on the computer using MP3decoding software.

Which one of us is right? Is there really a kind of compression that can extend the playing time of a disc in a conventional CD-audio player?

Felix Herzog via e-mail

Felix—I agree with you entirely. You cannot play MP3 files directly from a standard audio-CD player. Most such players can only read (play) files that are stored in Red Book audio format. The Red Book specification states, among other things, that the audio must consist of uncompressed 16-bit stereo files with a sampling rate of 44.1 kHz. Because MP3 files are compressed, your audio-CD player will not recognize them. Furthermore, the disc's lack of subcode data (used to identify tracks and timings, for example) will also cause playback problems.

As you stated, MP3 files are typically downloaded to a computer, where they can be decompressed and played back with a dedicated MP3 utility. You can also import the files into some audio-editing programs or export them to a portable MP3 player such as the Diamond Rio.

You can use Easy CD Creator to burn a CD-ROM of MP3 files, but your audio-CD player won't be able to read it. You'll have to load the CD-ROM into a computer drive and go from there. I hope this clarifies the situation.—David Rubin

#### **TOO MUCH TUBE TALK**

The resurgence of vacuumtube technology has given me some chuckles over the past few years. The hype is excessive. Tube gear is the answer to every social ill, it seems.

I learned electronics in the transition era between tubes and transistors. At that time, the goal was to reduce amplifier distortion, not to increase it. Get into your time machine and travel back to 1950. Congratulate an electrical engineer for achieving that "warm" tube sound. Explain to him that at the end of the century, engineers will like the distortion that tube amps add to the signal. His response: "You're accusing my amplifier of producing distortion? I worked two years on this design!"

Have you noticed that any product with the word "tube" on the label sells? As an engineer who has designed tube and solid-state amplifiers, I know that the tube is only one component in the aural equation. Also contributing to the tube sound are a poorly regulated (or unregulated) power supply, inherently nonlinear transformers, and circuit configura-

tions that can change the relative levels of even and odd harmonics. Solid-state designs lack these drawbacks (or benefits). Just putting a tube in the box does not re-create the audio nirvana of the 1960s.

If you like tube gear because it gives you a slightly distorted (warm) sound, then use it, buy it, love it! But let's tone down the hype and stop acting as though the old-timers planned all that distortion.

## Hank Wallace Fincastle, VA

Hank—The tube hype can be a bit much. I'll add that there are multiple ways to integrate a tube into an audio circuit, some more effective than others. A good tube processor can be a beautiful thing, especially when you're recording to a digital medium; but some so-called tube devices are better at looking cool than sounding warm.

As for the old-timers, they might not have wanted their circuits to distort the audio signal, but if they're honest they can't deny that some distortion occurs even in their cleanest tube circuits.—Steve O

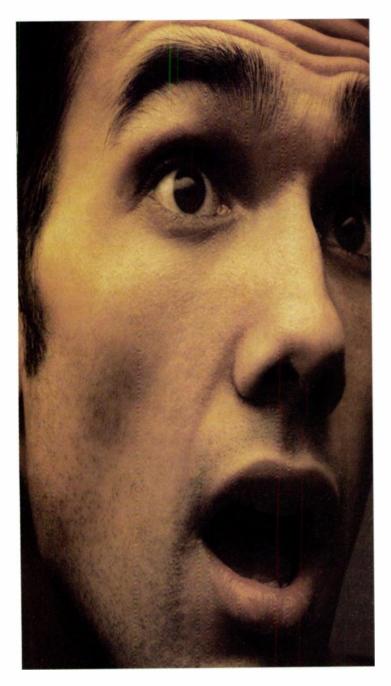
#### **SHARING SECRETS**

routinely record live acoustic music—mostly recitals, chamber music, and choral concerts. My work usually involves solo piano or piano and one other instrument, such as a violin. I'd like to share some tricks I've learned that produce great results every time. But first I have to say that I get to record impeccably maintained and

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#### LETTERS

tuned instruments played by impressive musicians in acoustically perfect halls and rooms.

I record everything in stereo using paired microphones. Under ideal acoustic conditions, an omnidirectional condenser mic yields the best response. But if the show is live and you want to minimize the coughing and talking that interrupts every performance, use cardioid mics facing away from the audience.

To get the most realistic recording of a piano or small ensemble playing classical music, use an XY stereo pair (I use Neumann KM 184s) on an extendable stand. Place them about ten feet up and angle them down at the performers. If you have more room and don't have a stereo-mic mount, use a spaced pair about ten feet apart. Always pan the pair far left and right-that's what stereo is all about! With a small ensemble, a spaced pair adds realism by capturing the location of each instrument in the space. Also, get good, heavy mic stands. I use the Shure S15A for my XY stereo work.

For solo piano, placing one mic over the bass strings and the other over the treble is a reliable all-around method for getting a good and very intimate sound from an unfamiliar piano. However, you do not get any information about the room around the piano if you use cardioid mics. Omnidirectional mics will solve that problem, and I also recommend removing the piano lid. (With a Steinway model D, on the other hand, I place the mics a few feet away because the piano needs more space.)

Another technique that I use is positioning two mics behind the pianist, one on each side. I find that the pianist has the best listening location. I place the mics about six feet up and angle them toward the piano. Take off the lid if possible so that it won't block the left-hand mic. This method results in an intimate sound but also captures the room behind the piano.

I've also had great success using two Audio-Technica AT4033s as a spaced pair. If you place the mics and set the levels correctly, you can often go straight to CD after editing only the beginning and ending of each piece.

I have a few suggestions about the audience. If 10,000 people are in the crowd, recording their response may give your final results more energy.

But if only ten people are there, edit out their clapping. Nothing minimizes a recorded performance more than pathetic, sparse applause.

> Michael Palmisano Cincinnati, OH

#### **VOCABULARY LESSON**

'm glad that you tried to explain the many acronyms found in today's music world [see "Desktop Musician: Alphabet Soup" in the February 2000 issue of EM]. But I was disappointed by the first definition that you offered.

You state that AC-3 (also Dolby Digital) is a "perceptual-encoding format that compresses six channels of digital audio in the now common '5.1' surround setup." This definition reinforces the common misunderstanding that AC-3 and 5.1 are inherently interrelated. They are not—5.1 is a channel format, and AC-3 is a method of data reduction.

AC-3 is an encoding method used to reduce the amount of data needed to store or transmit audio signals. You can use it on mono, stereo, and 3-, 4-, 5-, and 6-channel audio. But 5.1 is a channel format—a standard that describes the number of channels used, their frequency response, and the ideal placement of speakers in the listening environment [see "Square One: Surrounded by Sound" in the December 1999 issue of EM for more on 5.1].

AC-3 and 5.1 are completely separate items that can be used in conjunction with each other but do not need to be. Most important, AC-3 has nothing to do with creating a 5.1 mix.

Your definition might lead people to believe that AC-3 is somehow capable of turning any 6-channel mix into a 5.1 mix. AC-3 does not do this. The six channels (five main channels and one low-frequency effects channel) must already be in the 5.1 format before you can run them through the AC-3 encoding process.

#### Vance Galloway San Francisco, CA

Vance—You're on the money. People associate 5.1 with Dolby Digital because the two have been used together in the film world for the past several years and are the industry standard for creating and storing DVD video.—Larry the O

#### **TECHNICAL DIFFICULTIES**

n "The Electronic Century, Part II: Tales of the Tape" [see the March 2000 issue of EM], you stated that "there was no television in 1948." This could not be further from the truth. Television was in development in the late 1920s, was broadcast experimentally by CBS and NBC (and to some extent also by ABC and Dumont) in the 1930s, and made a public debut at the 1939 World's Fair. TV sets were available to the public from at least the late 1930s until 1942.

World War II put a kink in the technology's development, and TV was slow to recover after the war. But from 1946 through 1948, sales of TV sets rose steadily, and the Federal Communications Commission licensed more and more new stations. By 1948, TV service and sets were certainly available to most households, particularly in New York and Los Angeles—locations that each had seven actively broadcasting stations at that time.

In fact, 1948 was a banner year for television. The FCC was so swamped with licensing requests from new stations that it instituted a temporary moratorium on new-station licensing. It needed time to determine standards and to resolve reception-compatibility issues concerning black-and-white television, color television, and the sets available to the public. It was also concerned that stations in adjacent areas might cancel each other out. In 1952 the moratorium was lifted, but by that time millions of TV sets were in households across the country. The rest is history.

The author's original statement is clearly erroneous and, frankly, a rather glaring mistake.

Harry DeBusk via e-mail

#### **ERROR LOG**

April 2000, "Quick Picks," p. 180: The Big Briar Moogerfooger MF-102 ring modulator, not the MF-103 12-stage phaser, is pictured.

#### 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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## PRESONUS BLUETUBE

he PreSonus BlueTube (\$199) is a %U dual-channel microphone and instrument preamplifier featuring a 12AX7 tube and dual servo gain stages



without capacitors. The BlueTube can be used on its own or combined with the BlueMax stereo compressor to create a complete tracking system.

The BlueTube's front-panel controls are arranged in a "mirror image" configuration, with the inputs for its two channels on the far left and right sides. Each channel has Drive and Gain knobs, switches for a –20 dB pad and phase reversal, and an 8-segment LED meter for monitoring output gain. The channels share a 48V phantom-power switch.

The BlueTube's inputs use Neutrik combo connectors that accept balanced XLR and unbalanced ½-inch plugs. The rear panel has balanced XLR and unbalanced ¾-inch outputs; signal is available from both simultaneously. The BlueTube has an internal power supply and is designed to fit into a PreSonus BMRA rack adapter (\$29.95). PreSonus Audio Electronic; tel. (800) 750-0323; fax (225) 344-8881; e-mail presonus@presonus.com; Web www .presonus.com.

## KORG MS2000

org's MS2000 analog modeling synthesizer (keyboard, \$1,099; rackmount/desktop, \$799) combines the sound-creation capabilities of the company's vintage MS-series synthesizers with up-to-date DSP power and flexibility. The MS2000 offers 4-voice polyphony (last-note priority) and easy real-time access to most parameters via 35 dedicated front-panel knobs.

The MS2000 features two oscillators that draw from a palette that includes analog-modeled waveforms, DWGS-format (PCM) waveforms, and a noise generator. The 64 DWGS-format waveforms can be "wave sequenced" using the 3-track Mod Sequencer that also functions as an analog step sequencer. The Mod Sequencer can record parameter changes (including pitch) in real time or step time, and the recordings are editable.

You can use external audio as a waveform and process it with filters, effects,

> and other parameters. Filters modeled in the MS2000 include 12 and 24 dB lowpass filters, a 12 dB bandpass filter, and a highpass filter,



each with variable resonance that can be set into self-oscillation. The MS2000's arpeggiator can be synched via MIDI, and the two ADSR envelopes and 4-waveform LFOs can be synched to MIDI clocks, as can the Mod Sequencer, arpeggios, and delays.

Korg's Virtual Patching system emulates the hardware patching of the original MS synths. Envelope generators, LFOs, Velocity, and keyboard position can be used as modulation sources for pitch, noise level, and panning; eight sources and eight destinations can be selected from the front panel.

The MS2000 includes a 16-band vocoder (modeled after Korg's VC-10) and a three-part effects section with modulation effects, delay, and a 2-band EQ. Korg USA; tel. (516) 333-9100; fax (516) 333-9108; e-mail product\_support@korgusa.com; Web www.korg.com.

### 🖊 KRK VA

RK's V6 active near-field monitors (\$799 a pair) are intended for use in facilities with strict space requirements, such as remote trucks, DAW environments, and small studios. Users wishing to employ them in a surround system can take advantage of KRK's free Orbital Surround Solutions option, which matches all speakers to within a ±0.5 dB tolerance.

The monitors are made of 0.75-inch medium-density fiberboard with a 0.5-inch port shelf and an extrathick front baffle, and they come standard with video shielding. Each speaker has two separate power amplifiers and an electronic crossover that matches power output

to frequency for each driver. The 6-inch polyvinyl woofer and the 1-inch silk dome tweeter are powered by 60 and 30 watts, respectively. The woofer features a new antiresonant frame designed to minimize the ringing effect associated with metal frames.

Neutrik combo inputs on the back of the speakers accept either balanced XLR or balanced ½-inch TRS connec-

tors. A System Gain control

allows variable input sensitivity from +6 dB to -30 dB. Frequency response is rated at 54 Hz to 20 kHz, with a maximum SPL of 102 dB for music, 105 dB for peaks. Two V6 speakers weigh a total of 59 pounds. KRK Systems; tel. (714) 373-4600; fax (714) 373-0421; e-mail sales@krksys.com; Web www .krksys.com.



## CREAMWARE POWERSAMPLER

reamWare's PowerSampler (\$598) is an integrated sampling system for Windows and the Macintosh OS that offers an alternative to latency-laden and CPU-intensive software samplers. PowerSampler is multitimbral, with 16 channels and a maximum of 32 dynamically allocated stereo voices. The PowerSampler PCI card includes three Analog Devices SHARC DSP chips that handle the sampling, mixing, and effects duties. The host CPU deals with sequencing, hard disk recording, and software-synthesis chores.

PowerSampler loads samples into the host CPU's RAM; within the limits of your CPU, it will address as much RAM as you like. Files in Akai S1000 and 3000, Sound-Font 2.0, WAV, and AIFF formats are sup-

ported. You edit them with the integrated graphic sample-editing software.

You can send each channel to the main stereo outputs or any of the six individual outputs. These outputs are routed to a built-in 24-channel mixer, where two aux effects can be applied.

The PCI card features stereo analog and S/PDIF I/O, which supports 24-bit, 96 kHz sampling and playback. The S-Link interface allows you to connect the card to the optional breakout box (price TBA) for eight additional analog inputs and outputs. You can also attach a 16-channel ADAT interface to the card. A second PCI card can be used via the S/TDM Bus Connector to double your polyphony and I/O.

PowerSampler boasts a latency of only 1 to 2 milliseconds and includes drivers

for ASIO, EASI, MME, DirectSound, OMS, and the NemeSys GigaSampler. CreamWare US; tel. (800) 899-1939 or (604) 435-0540; fax (604) 435-9937; e-mail info@creamware.com; Web www .creamware.com.



## **Z** ARBORETUM REALIZER PRO

esigned to improve the sound quality of Internet audio, Arboretum's Realizer Pro plug-in (\$74.95) is a comprehensive cross-platform authoring tool for enhancing low-bandwidth Internet audio. Realizer Pro supports DirectX (Windows) and VST (Macintosh)



formats. In addition to MP3 files, *Realizer Pro* can enhance files in the AIFF, WAV, SDII, and other audio formats.

Adapted in part from Arboretum's Hyperprism suite of sound-design tools, Realizer Pro includes a number of useful effects: Bass Maximizer synthesizes harmonics to improve a file's bass response, Harmonic Exciter synthesizes partials in the upper frequency range, Ultra Stereo widens the stereo image, and Max Loud normalizes a file and evens out its dynamic range. The plug-in also has a two-stage parametric EQ. Arboretum Systems; tel. (800) 700-7390 or (650) 738-4750; fax (650) 738-5699; e-mail info@arboretum.com; Web www.arboretum.com.

### **Y** ECHO MONA

he newest release from Echo is Mona 24/96 (\$995), a 24-bit, 96 kHz digital-recording package that includes a PCI card, an outboard I/O unit, and a software driver. Mona offers S/PDIF and ADAT digital I/O, and its inputs and outputs provide 115 dB of dynamic range.

Mona's four analog inputs use balanced XLR/%-inch combo jacks that accept instrument-, mic-, and line-level signals. The gain range of the trim pot automatically changes according to the type of input, and each channel has an input meter. The mic preamps offer globally switchable phantom power.

Six analog outputs are available on either XLR (+4 dBu) or RCA (-10 dBV) connectors. The optical port can be used for S/PDIF or 8-channel ADAT Lightpipe I/O. S/PDIF I/O is also available on coaxial jacks. For synchronization, you get

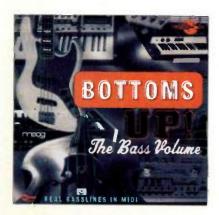
word-clock I/O and a 9-pin connector that allows Mona to communicate with other Echo 24-bit products by means of its proprietary Esync protocol.

Analog and digital I/O can be used simultaneously, allowing a maximum of 26 tracks (4 analog in, 6 analog out, 8 ADAT in, 8 ADAT out) of recording and playback. Channels 1 and 2 can be monitored via the headphone jack (with a dedicated volume control) on the front panel.

Mona is compatible with most editing/recording software and comes with a custom version of Syntrillium's *Cool Edit Pro* (Windows 95/98). PC users will need a Pentium system with 16 MB of RAM, Windows 95/98/NT, and a PCI 2.1 slot. Mac users will need a 604 processor, OS 8.1, and as much RAM as their software requires. Mona also runs on BeOS 5.0. Echo Digital Audio; tel. (805) 684-4593; fax (805) 684-6628; e-mail info@echoaudio.com; Web www.echoaudio.com.



## SOUND ADVICE A A A



## ▲ KEYFAX

7 eyfax Software continues its tradition of authentic-sounding Stan-Adard MIDI Files with the release of Bottoms Up (\$29.95) and Hip Hop Shop (\$39.95). Featuring bass lines recorded by top professionals on alternate MIDI controllers, Bottoms Up contains parts suitable for a wide variety of instruments, including electric and acoustic bass, synth bass, plano. and tuba. The collection includes 700 grooves in styles ranging from funk, alternative, modern rock, and R&B to hip-hop, reggae, Latin, and jazz, the latter with lots of walking bass lines.

The performers on Bottoms Up have impressive credentials. Among them are Tiran Porter (Doobie Brothers),

Paul Jones (Whitney Houston), Phil Retamoza (Orquesta Gitano), Dennis Murphy (Acoustic Alchemy), and Julian Crampton (George Benson).

Hip Hop Shop is a collection of hiphop and R&B grooves produced by Ron Beck (Tower of Power, Sister Monica), Al Eaton (Ice T, Queen Latifah), the Beat Professor (Chronic Music), and "Pharmacyst" Bill Turner. The contents include drum loops. bass lines, keys, and effects, all arranged by category for easy use with any MIDI sequencer. Keyfax Software/Hardware; tel. (800) 752-2780 or (831) 460-0172; fax (831) 460-0173; e-mail us@keyfax.com; Web www.keyfax.com.

## **NEMESYS**

aking advantage of the huge sample capacity of GigaSampler, NemeSys has released Jim Corrigan's Nashville High-Strung Guitars (\$299). High-strung guitars are guitars with the three lowest strings tuned an octave higher than normal in order to create sonic space in a mix for other instruments with similar overtone structures. Highstrung guitars are usually strummed for rhythm guitar parts and impart a unique sound to a song.

This collection of samples in the GigaSampler format allows access to both up and down chord strums in every key (with dynamically variable chording), as well as single notes with full decays and performance articulations. The samples were recorded on a six-string dreadnought guitar.

NemeSys has also introduced Conexant General MIDI Libraries



(GM150, \$79; GM500, \$149), a pair of General MIDI sample-set CD-ROMs for GigaSampler. The GM150 library is a 150 MB sound set, while the GM500 is a 500 MB sound set. Neme-Sys Music Technology; tel. (512) 219-9181; fax (512) 219-9029; Web www .nemesysmusic.com.

## CM AUTOMATION PM 64

The CM Automation PM 64 router/ level controller (\$2,500) is a 2U automated patch bay and level-matching system designed for recording, broadcast, and sound reinforcement and installation applications. The

PM 64 has a 32-by-32 routing matrix that handles 32 mono or 16 stereo inputs and 32 mono or 16 stereo outputs. The unit can also add scene and dynamiclevel automation to mix-

ers when used with a MIDI sequencer.

The PM 64 can store 90 setups in nonvolatile memory. Setups include all audiorouting and level settings, 16-channel MIDI patch maps (for sending patchchange commands), and internal input and output "labels." The PM 64's front panel has 32 Source and 32 Destination switches for selecting audio channels.

> You use the Route, Level, Meter, and Save switches to choose operations, and a knob to select patches, adjust volume,

and modify other parameters. A 12-LED meter (+4 dBV/-10 dBu, switchable) displays input or output levels over a range of -30 to +20 dB for any selected channel. Audio inputs and outputs are on eight gold-plated DB25 female connectors.

> The PM 64's frequency response is rated at 2 Hz to 120 kHz, with a 119 dB dynamic range, a -92 dBV noise floor, and 0.004% THD+N at 1 kHz (20 Hz to 20 kHz). CM Automation; tel. (888) LUV-MIDI or (818)

709-4732; fax (818) 709-4039; e-mail pmontes@cmautomation.com; Web www .cmautomation.com.

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# KEY

Steinberg North America has announced that it will be a distributor for Native Instruments, a producer of software-based synthesizer systems such as Reaktor and Pro 5... Ilio Entertainments will begin distributing Sonic Reality sample libraries . . . Line 6 users can now download new sounds gratis via the Tone Transfer Web Library at www.line6.com. A variety of criteria allows users to search by artist, song title, style, or amp model . . . Alesis Sound Bridge 3.0 is up at www .alesis.com. For the companion video tutorial (\$29.95) call (800) 5-ALESIS or e-mail parts@alesis.com ... Waves now offers complete support for Digidesign's Pro Tools LE 5.0 software, with its WaveShell-RTAS, which provides support for Digidesign's Realtime AudioSuite format ... TC Works has released a new TDM plug-in version of its Spark stereo audio-editing software (\$699). The new TDM version offers denoising and declicking, features that are not available in the stand-alone Mac version. The company also has ported its MegaReverb TDM plug-in (\$799) to the Windows NT version of Pro Tools ... CreamWare has released its Pulsar DSP system for the Mac OS (\$1,398 at press time). Crossgrades between Windows and Mac versions are free . . . E-mu announced that it has solved a chip problem, putting its Ensoniq product line back on track-including new versions of the ASR-X sampler, ZR-76 keyboard workstation, and Paris digital audio workstation . . . Sounds Logical has released its WaveWarp 1.2 (\$199), which features multichannel, realtime audio inputs. Buy it at www .soundslogical .com. A Lite version is priced at \$99 . . . SEK'D America announced the integration of its Comparisonics technology into Samplitude 2496 version 5.5.



## STEINBERG REASON

complete self-contained package that can be used in conjunction with any MIDI sequencer, Steinberg's Reason (\$399) is a software studio in a box. It includes a synthesizer, a sampler, a drum machine, a REX-file player, effects, and a sequencer. In addition, its ReFill format allows sounds, loops, patches, and sequences to be bundled and published on the Web or sent to another Reason user.

Included in *Reason* is SubTractor, an analog modeling synthesizer; the NN19 Digital Sampler; and the Redrum Drum Machine. You can process these modules using a variety of effects, including reverb, delay, phaser, chorus, flange, compressor, EQ, envelope filter, and distortion. The effects processors can be modulated via MIDI or a virtual control voltage (CV).

The Matrix Pattern Sequencer emulates a classic analog step sequencer with its ability to send virtual gate and CV output to other *Reason* devices. The MTG-128 MIDI-to-Gate Converter translates incoming MIDI messages to virtual gate signals for use in other *Reason* devices as well.

The Reason Sequencer allows graphic editing of all the events, customized to the type of *Reason* device selected. The package also includes a ReWire device that lets you connect other ReWirecompatible programs (such as *ReBirth*) to *Reason*'s mixer.

Reason comes with a database and browser for managing the files associated with the program, such as samples, patches, loops, and MIDI data. The software supports ASIO, MME, DirectX, and Sound Manager. It can import 24-bit files and export AIFF and WAV files, as well as MP3s (using a helper application). Reason is bundled with more than 500 MB of samples, loops, drum kits, and patches. Steinberg North America; tel. (818) 678-5100; fax (818) 678-5199; e-mail info@steinberg-na.com; Web www.cubase.net or www.us.steinberg.net.

## LINE 6 POD PRO

ine 6's Pod Pro (\$799.99) is a 2U version of the original Pod guitar processor that boasts additional features and expanded I/O. The unit features 32 amp and 15 cabinet models, 15 effects combinations, and 36 user-memory locations for storing custom setups.

Pod Pro has guitar and line-level inputs, and its XLR outputs allow you to send both direct and processed

signals simultaneously. Digital output is 24-bit at either 44.1 or 48 kHz, in both S/PDIF and AES/

EBU format. Word-clock I/O is also included. The handy ToneTransfer feature lets you transfer sounds between the Pod Pro and other Line 6 products (including the original Pod and the Flextone II amp), or download them from the company's Web library.

Line 6 has also released the Bass Pod (\$499.99), which models 15 cabinets and 16 classic bass amps (including Ampeg SVT and B-15, Marshall Major and "plexi" Super Bass, SWR, Versatone, and Vox AC 100), along with effects tailored for use with bass. The Bass Pod allows simultaneous output of direct and processed signals (a headphone output is included); its ToneTransfer feature



enables sounds to be transferred between Bass Pods and downloaded from the Web library. Line 6; tel. (805) 379-8900; fax (805) 381-4684; marketing/R&D fax (805) 379-3001; e-mail info@line6.com; Web www.line6.com.

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Walli

Walli Rack 24 Power Pack (evallable for both PC and MAC)



We have been thinking that 4 Mic preAmp, 8 analog audio channel, A/D/A with 120dB dynamic range 4In/4Out MIDI interface, Word Clock I/O, SMPTE I/O, 24-bit, 96kHz....are not enough. So, we decide to include software....

WaMi Rack 24 with Smaplitude Project v.5.5 and special Gigasampler LE(8ch/48kHz) At only \$789.00



Mac OS





Compatible with all major recording software

## SAMPLITUDE PROJECT

SYMPELIONE Project 3:50 v 2 3

ult or So his teated features,
is held strictly for gower users!
in him atom XIII be your machination repeated feating your held to according a confine of SP mixing / M DI agreements software from SEKD (\*for PC)

## GIGASAMPL

**Waveterminal 2498 Power Pack Evallable for both PC and MAC** 

It's just right time for you to get the Waveterminal 2496. 24-bit, 96kHz super sound quality audio interface meets Powerful software package.

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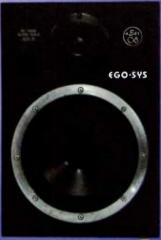
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Total of 18 digital charnes with dual ADAT ports and splatforal 24-bit. 96KHz digital I/O (AES/EBU or S/PDIF) gives you enough room for creation.

2 ADA; Tos ink Input & Output(24-bit 16 channels), S/PDIF Coaxial digital Input & Output. AES/EBU XLR digital Input & Output. Total 18 ch of Digital I/O (Ready for 24-bit 96 kHz), 54 channel Internal Digital Mixer for monitoring, BNC type Word Clock Input & Output, 2 MDI In and 2 MIDI Out ports(32 MIDI channels), 9-pin ADAT sync In connector, Inclided 20-bit Analog Bracket for analog monitoring, Windows 95/98/NT4.0/2000 Ready!







## Surprisel

High quality studiu reference monitor system. You can't stop once you turn it on:

## Audiotrak 2000



6 analog Jine inputs & 8 analog line outputs, A pair of Mic PreAmp with Phantom Power, A pair of Headphone Amps with level control, 8 of 20-bit DAC, S/PDIF (Coaxial) digital Output, Full-Duplex - All Channels Record & Playback simultaneously, High Performance RISG-based Audio DSP (no CPU dependent) provides International Mixer & Professional effects Processor(Reverb, Delay, Chorus, Echo, EQ. etc.), 64-voice multi-timbral hardware wavetable Synth/Sampler 16 Mb on-board Sample RAM(expandable up to 64MB), 1 Input Port and 1 Output Port MPU-401 MIDI Interface

Time to clean up your working space. Looks small but carries huge capability





## WaMi Box

PCMCIA digital audio/MIDI system built in 64-voice synth/sampler



## Waveterminal U24

USB-Quench the thirst 24-bit ready audio interface for USB. Enough to say.



## Miditerminal M4U

Takes all the function from 4140 and more.... It is made for USB!



## Dr. D

Ultimate solution for digital signal converting. Finally, they can understand each other.



## Miditerminal 4140

4in 4Out MIDI Interface supplying more I/O and more stability and even SMPTE Generator/Reader

## PERSONAL DIGITAL STUDIOS A A A



## YAMAHA AW4416

he AW4416 Professional Audio Workstation (\$3,299) by Yamaha integrates a 16-track hard disk recorder, an 02R-like automated digital mixer, extensive audio processing, a stereo mastering system, and a sampler into a desktop-size package. The unit ships with a 10 GB hard drive that yields up to 80 minutes of 24-bit recording. Storage is expandable up to 64 GB. For additional recording and playback, consider the optional onboard CD-RW drive (\$500).

The AW4416 records 16- or 24-bit, 44.1 or 48 kHz audio without compression. Eight tracks can be simultaneously recorded, and 16 tracks simultaneously played back. The All Rec mode allows you to record 16 tracks at once (at either resolution), albeit with reduced DSP functions. The AW4416 provides 128 virtual tracks and a 2-track stereo mastering section, for a total of 130 internal tracks.

Channels 1 through 8 have ¼-inch TRS analog mic/line inputs and analog trim pots. Channels 1 and 2 include phantom-powered XLR mic inputs and ¼-inch TRS insert points, while channel 8 has a high-impedance, ¼-inch guitar input. You also get stereo coaxial S/PDIF digital inputs and outputs. Analog outputs include stereo pairs on RCA and balanced ¼-inch connectors, as well as four balanced ¼-inch multipurpose outputs.

The stock I/O can be expanded by

adding up to two optional mini-YGDAI cards. Available card configurations include eight channels of ADAT or TDIF I/O, eight channels of AES/EBU inputs on four stereo connectors, eight channels of balanced ¼-inch analog inputs, four channels of balanced XLR inputs, and four channels of balanced XLR outputs.

The fully automated mixer sec-

tion boasts 32-bit processing, a serial mouse port, 17 motorized 60 mm faders, a 320-by-240-pixel backlit display, two multi-effects processors, and a total of 20 buses that can be routed to various outputs. Individual channels and all buses feature 4-band EQ, dynamics processing, delay, attenuation, pan, and phase reverse. There is an onboard stereo mastering section, with mixdown to either an external recorder or discrete internal stereo tracks. An onboard sampler with 16 trigger pads and 90 seconds of dynam-

Frequency response is rated at 20 Hz to 20 kHz (+1/-3 dB), with a typical dynamic range of 105 dB and <0.02% (at 1 kHz) total harmonic distortion. Yamaha Corporation of America; tel. (714) 522-9011; fax (714) 739-2680; e-mail info@yamaha.com; Web www.yamaha.com.

ically allocated memory completes the

## AKAI DPS16

picture.

kai's new DPS16 (\$2,695) offers 16 tracks of uncompressed 24-bit

A digital recording at four sampling rates (32, 44.1, 48, and 96 kHz), 56-bit internal processing, a 26-channel digital mixer, and onboard waveform-level editing capabilities. The DPS16 comes with a 10 GB internal IDE/SCS1 drive; a SCS1 port allows connection to as many as seven additional external SCSI devices. CD-R

and CD-RW drives are also supported.

The DPS16 can record 10 tracks (8 tracks at 96 kHz) simultaneously and play back 16 tracks at a time, with a total of 250 virtual tracks available. Recording time and frequency response are determined by sampling rate. For example, you get 56 minutes of 8-track recording at 24-bit/96 kHz, with a frequency response of 10 Hz to 44 kHz, or 2 hours of 16-track recording at 24-bit/44.1 kHz, with a frequency response of 10 Hz to 20 kHz.

The 26-channel mixer features 24-bit/ 96 kHz delta-sigma converters with 128× oversampling. Channels 1 and 2 have combo jacks that accept either balanced XLR or balanced ¼-inch TRS connectors, with switchable phantom power on the XLRs. Channels 3 through 8 have balanced/unbalanced ¼-inch jacks, and channel 8's jack is switchable to high impedance. Outputs include four ¼-inch balanced aux sends, as well as stereo master and monitor outputs on RCA connectors.

The mixer has 16 faders, and all input channels have trim and pan pots, four aux sends, and 3-band EQ with sweepable high and low bands and fully parametric mids. Snapshot-scene memories can be stored internally, and dynamic fader automation is possible using an outboard MIDI sequencer. You also get four channels of onboard effects, including pitch correction, a vocoder, reverbs, and delays.

Recorder and mixer functions are displayed on a flip-up 320-by-240-pixel





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## PERSONAL DIGITAL STUDIOS 🔺 🔺 🔺

backlit LCD. The Q-Link Navigation System offers real-time control over all major functions and screen parameters via six virtual Q knobs located just to the right of the LCD. Graphic waveform editing, with sample-accurate resolution, allows you to edit and move single or multiple tracks to and from anywhere within the recorder, and the device has 256 levels of undo. Akai's Timestretch function provides 54 preset algorithms for phase-coherent processing of stereo signals.

The DPS16's dynamic range is rated at >100 dB; its distortion is <0.003% (at 1 kHz, when sampling at 44.1 kHz). Akai Musical Instrument Corporation; tel. (800) 433-5627 or (817) 831-9203; fax (817) 222-1490; e-mail info@akaipro.com; Web www.akaipro.com.

## ► ROLAND VS-1880

ollowing hot on the heels of its popular VS-1680, Roland's VS-1880 (\$2,795) adds new 24-bit converters, track assignments that are more flexible, improved CD mastering capabilities, and other upgrades. When in the MT Pro recording mode, the new converters maintain a true 24-bit resolution from input to output.

The VS-1880 records 8 tracks and

can play back 18 tracks simultaneously, with 288 virtual tracks available. You can use any 2 of the 18 tracks for 2-track mastering and CD burning. The Audio Image Format feature, linked to the CD-RW Mastering button, simplifies and accelerates the burning process by giving faster access to the CD-writing menus

and allowing you to save image files to disk. The CD features are designed for use with the optional CD Recording System (\$750).

The VS-1880 includes ten 24-bit audio inputs. Channels 1 and 2 have balanced XLR mic connectors with phantom power; channels 3 through 8 have balanced ¼-inch TRS connectors; and channel 8 has an additional high-impedance guitar input. Stereo S/PDIF digital I/O is available on optical and coaxial connectors. Analog outputs are via eight unbalanced RCA connectors, and a stereo headphone jack is included. A SCSI interface and MIDI In, Out, and Thru jacks complete the list of connections.

The fully automated 28-channel mixer allows simultaneous mixing of the 18 recorded tracks and 10 inputs.



The centerpiece of the user interface is a 320-by-240-pixel LCD and 13 motorized faders. You have a choice of using either 3-band EQ on 18 channels or 2-band EQ on 28 channels.

There are no onboard effects, but the VS-1880 accepts two optional VS8F-2 Effect Expansion Boards (\$300). Each adds four stereo or eight mono effects to complement Roland's exclusive COSM-based modeling effects (such as mic, guitar amp, and speaker models), as well as the Mastering Tool Kit, which provides a multiband compressor and other mastering processors.

The VS-1880 weighs a little over 13 pounds and comes bundled with Emagic's *Logic VS* and Liquid Audio's *Liquifier Pro* software. Roland Corporation U.S.; tel. (323) 890-3700; fax (323) 890-3701; Web www.rolandus.com.

## ► TASCAM MX-2424

he Tascam MX-2424 (\$3,999) is a 24-bit digital hard disk recorder that records 24 tracks at 48 or 44.1 kHz, or 12 tracks at 96 kHz. (The 88.2 kHz sample rate is also supported.) A 9 GB internal SCSI hard drive accommodates up to 45 minutes of recording time; additional drives can be installed in a 5.5-inch drive bay on the device's

5.5-inch drive bay on the device's front panel and via a port on the rear panel.

The MX-2424 comes with the ViewNet MX Graphic User Interface and has built-in editing capabilities that include 100 levels of undo and 999 virtual tracks.

The recorder reads and writes files in WAV and Sound Designer II formats. Onboard metering for all 24 channels is standard.

Stereo AES/EBU (XLR) and S/PDIF (coaxial) digital I/O is included; all other I/O is available on optional cards. The MX-2424 recorder has one digital

THOSE WE SEE THE SECOND SECOND

card slot that accepts 24-channel cards for TDIF (\$499), ADAT (\$499), and AES/EBU (\$999) connectors; a second slot accepts an analog I/O card (\$1,499) that employs 24-bit, 96 kHz converters with I/O via six DB-25 connectors. The analog and digital cards can be used simultaneously.

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How much would you expect to pay for a mic pre with all these features? A thousand? Eight hundred? Less than six hundred? You're getting warmer.

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## WebPage

By Gino Robair

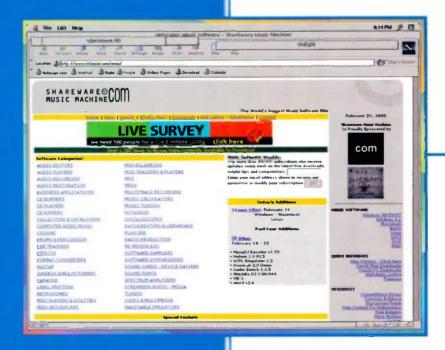
## DOTDOTDOT.COM

## WEB SITE OF THE MONTH

Shareware Music Machine (www .hitsquad.com/smm) touts itself as the "world's biggest music software site." Among its whopping 44 categories of downloadable music-related applications are Audio Restoration, CD Rippers, Label Printing, MIDI Players and Utilities, Metronomes, Oscilloscopes, and Video and Multimedia. You can browse the categories on any of the eight available platforms (including Linux, BeOS, and Atari), check out the Top 20 Mac or

PC downloads, and peruse the message board. The site also includes a reeware page for your favorite computer platform.

A multitude of interesting applications is available at this site. I had a fantastic time scouting out programs for my system, and I had just as much fun looking at applications for platforms I don't even use. A word of caution, though: when you log on to Shareware Music Machine, prepare to stay awhile.



its Lava Producer application (\$29.95), available online at www.lava.com. You can use Lava Producer in conjunction with the company's free Lava Player-an interactive video environment with 3-D images that move and morph in sync with an MP3 file-to customize MusicVideo 3-D (MV3) files incorporating your own visual material. For example, you can take pictures with a Creative Labs Video Blaster WebCam and drop them onto the objects in one of Lava Producer's screen templates. All of this can be done in real time. The minimum system requirements are a Pentium II/233 MHz running Windows 95 or 98, 32 MB of RAM, a sound card, and an OpenGL graphics accelerator card . . . EarBuzz (www .earbuzz.com) is a new e-commerce Web site where independent musicians can market their CDs and other artist-related merchandise, such as T-shirts, hats, and stickers, EarBuzz provides artists with a number of free items, including a credit-card transaction service, a personal page on the site, links to an artist's own Web site, and online promotional opportunities. EarBuzz's contract is nonexclusive, so musicians can take advantage of other avenues of distribution at the same time . . . PG Music (www.pgmusic.com) has added a new chat feature to its Web site so that users can get one-on-one customer service and technical support. Just go to the site and click on the Live Chat button to contact a service representative. Hours of operation are 6 a.m. to

9 p.m. PST, seven days a week.

Creative Labs has just introduced



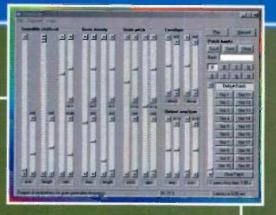
## DOWNLOAD OF THE MONTH

If you haven't experienced the fascinating sounds of granular synthesis, now's your chance. GranuLab 1.0 (Win) (http://hem.passagen.se/rasmuse/Granny.htm) is a freeware granular-synthesizer application by Rasmus Ekman that allows you to process audio files or create sounds from scratch.

A granular synthesizer produces its unique sound by playing large groups of tiny sound fragments, or grains. Each grain can be as simple as

a sine wave or as harmonically complex as a sample. Each of the grains can have its own set of parameters, many of which can be adjusted in real time. With granular-synthesis techniques, you can easily create rich and complex sounds.

GranuLab gives you real-time control over the pitch, envelope, and density of the grains. It allows you to apply interesting time-stretching and pitch effects to a sample while retaining the character of the original sound. Parameter configurations can be saved as patches, and you can morph between up to four patches at a time using your mouse as a controller. The application also has MIDI-controllab!e



features. A word of caution from EM associate editor Dennis Miller: be careful not to set the grain density too high, lest you overload your CPU.

If you want to work at a higher level, you can move up to *GranuLab 8*, a shareware application that lets you run up to eight separate patches at once. *GranuLab 8* has a number of additional features, such as panning and the ability to load new files while the program is running.



#### **Got Melk?**

Anyone who has tried to land a major-label deal knows that artist visibility is an important factor. So how do you make yourself heard above the noise of a million bands vying for A&R attention?

A Garageband.com contest (www.garageband.com) awards a \$250,000 recording contract to the unsigned artist or band that receives the best visitor response on the site. The British band Melk occupied the number one spot for more than three weeks with She's Been Sleeping, its



second release on the Garageband site.

Melk was formed in 1997 when, following the release of her solo album *Tundra* (Sony, 1994), vocalist Sara Davis collaborated with engineer/producer Neil Simons on a remix of one of her songs. Afterward the two wanted to continue their working relationship but in a different musical direction. Together, Davis and Simons created the flavorful hybrid of smooth, folklike vocals and electronic beats and processing that gives Melk its unusual sound.

Davis and Simons spent two years writing songs and building a following in London. They added a rotating lineup of guest DJs, hip-hop poets, and other musicians to their performances to augment their lazy grooves and luscious melodies.

The members of Melk are no strangers to the push-pull world of the record industry. After a couple of years of flirting with several labels, the band took matters into its own hands. "Garageband.com was the obvious place for us to put our songs,"

says Davis. "It gave us instant feedback from our musical peers, with no A&R men telling us to add more bass. It's a great solution and loads of fun."

Davis's relocation to the San Francisco Bay Area hasn't slowed the group down. Melk now records over the Internet using the technology provided by Rocket Network (see "Web Page: We Have Liftoff!" in the April 2000 issue of EM). "I usually post an acoustic track to our private Internet recording studio at 8 p.m. West Coast time," Davis explains. "The next morning, I log on and listen to what Neil added while I was sleeping."

The partners feel that the success they've had on Garageband.com is just the beginning. "Bands on the Internet can reach more people in a week than they can in two years of touring," says Davis, "at virtually no cost."

"As the Internet is becoming the 'Enternet,'" she continues, "we're positioning ourselves to capitalize on all our hard work by reaching the audience on our own terms. This is every band's dream!"



## WEBCAST

Digevent (www .digevent.com), a company specializing in interactive Webcasts, has entered the field of online music education. Digevent offers free weekly online lessons, held regularly on a specific day and time, to anyone who is interested. The maximum class size is 2,500 students, and so far Digevent has been able to accommodate every applicant.

The current course offerings include Drum Beat, Keyboard Kaleidoscope, and Guitar Mania. Each class meeting is divided into segments. For example, a Guitar Mania lesson devotes time to a Riff of the Week discussion and a question-and-answer period. On occasion, a guest artist will focus on a specific topic.

You register for classes online at the Digevent Web site; the process takes about 15 minutes. During registration, the Digevent server checks your computer to make sure that you have the applications you need to participate in the Webcast. (The company refers to this as *preflighting*.) The minimum system requirements are a Pentium/166 MHz running Windows 95, 98, 2000, or NT; an Internet connection with a 56 Kbps modem; and Microsoft's *Media Player* 6.4 and *Internet Explorer* 4.0.

After you've registered, Digevent e-mails you the lesson site's URL and your log-on time. During a lesson, you can e-mail questions to the instructor. And if you miss a class, you can catch up by downloading an archived copy of the lesson.

Digevent plans to make the service accessible to Macintosh users later this year. Master-level classes, available on a pay-per-view basis, are also in the works.



### **Creating MP3 Files**

The MP3 (short for MPEG-1 Layer III) compression protocol is the most popular Web-delivery format for music. The MPEG algorithm dramatically reduces audio-file size by using Fast Fourier Transforms to divide a track's frequency spectrum into discrete bands. The MP3 encoding process employs psychoacoustic principles to reduce file size with minimal effect on the frequencies heard by the human ear.

To create an MP3 file, you need an encoding program, often referred to as a ripper. A variety of shareware and commercial applications are available on the Web, but I usually use AudioCatalyst from Xing Technology (www.xingtech.com) and the PC version of WaveConvert from Waves (www.waves.com) with an MPEG-encoding plug-in installed. QDesign (www.qdesign.com) makes an especially efficient codec for professional use.

WaveConvert is the Swiss Army knife of audio-conversion utilities. AudioCatalyst is more specific, geared toward pulling songs off a CD and putting them on your hard drive in MP3 format. Both programs allow you to select from a range of output bit rates—the higher the rate, the better the audio quality (and the larger the file). The MPEG codec is notorious for smearing sharp transients, such as the notes from a snare drum. Consequently, a rock tune is best encoded at a higher bit rate. On the other hand, you could successfully use a lower bit rate on a recording of a choir.

The standard bit rate for Red Book (16-bit, 44.1 kHz stereo) audio is 128 Kbps, which results in about a 10:1 compression ratio. If you start with a mono file and limit the bandwidth a little with an EQ, you can use lower bit rates without generating audio artifacts (often heard as electronic bubbling noises). Voice-overs, with their small frequency range and lack of hard attacks, can usually be compressed at 32 Kbps or lower, resulting in an amazing 20:1 or better ratio. AudioCatalyst also features a "variable bit-rate" format that will, for example, encode the sounds of a car crash at a high rate and the ensuing silence at a lower rate in the same file. This efficient scheme results in a consistently better sound, but the format is not

supported by all MPEG players.

An MP3 file can be posted on a Web site for download, but its behavior depends on the user's platform and plug-ins. On the PC, Microsoft's Internet Explorer automatically streams any MP3 file it encounters, as does the Apple QuickTime 4 plug-in. Setting up streaming on a Mac using Netscape's Navigator browser is a little more problematic. On my system (a Power Mac G3/300 MHz running OS 8.5 and using a T1 line for Internet access), I have to download an entire file and then play it using a "helper" application.

Some Web software, such as *Beatnik* and Macromedia *Flash* 4, have built-in MPEG-compression utilities. These keep the output files small and allow several low-bandwidth delivery methods, including streaming and interactivity. For more information, check out the MPEG Pointers and Resources page at www.mpeg.org.—*Peter Drescher* \*



# M.ONE D.TWO

DUAL EFFECTS PROCESSOR

MULTITAP RHYTHM DELAY



## STATE-OF-ART-SOUND

Behind the cool looking exterior of the M-One and D-Two Effects Processors you'll find more than 20 years of know-how and experience in creating high quality sound machines with state-of-the-art sound. These Ultimate Sound Machines combine cutting edge technology with affordability and they deliver a wide range of high quality effects!

# THE DUAL EFFECTS PROCESSIN



## **CUTTING EDGE TECHNOLOGY**

The M-One, Dual Engine Multi-effects processor, will revolutionize your work with effects. It boasts 24 bit S/PDIF digital I/O and balanced analogue inputs and outputs. The M-One has a Dual Engine structure, 1/4" balanced jack I/Os, 44.1-48kHz internal processing, various routing options, high quality reverb algorithms as well as other useful effects and processes.

## HIGH QUALITY ALGORITHMS

You get more than 20 high quality reverbs from the classic Halls and Rooms to new and grainy snare reverbs such as Live and Plate. The M-One also gives you various processing algorithms such as Delay, Chorus, Phasing, Flanging, Tremolo, Pitch-shifting and Detuning as well as Parametric equalization and a wide range of dynamics tools. Use the M-One's many high quality reverbs to create sound reflections in various environments.

## INTUITIVE USER INTERFACE

Now you can make music instead of wasting time programming! TC's world acclaimed intuitive user interface will take you there fast and easy. Adjusting an effect is just as simple as pressing the ALGO/EDIT button for the relevant Engine and stepping through the available parameters with the cursor buttons.

## **FLEXIBLE ROUTINGS**

The Dual Engine's ru 'ure of the M-One gives you flexible routing options of the two Engines. Use the M-One as a true stereo unit with the Engines performing identical tasks on the two channels, a

Dual Mono mode, a Dual Send/Return mode with common stereo return, a separate Parallel processing mode with stereo I/O and separate Serial processing mode also with stereo I/O.

In the Dual Send/Return mode the two Engines remain entirely separate.



The flexible routings enable you to run two of the best sounding Reverbs or other quality effects simultaneously without loosing the power and integrity of your original tone.

### PRESETS

The M-One comes loaded with 100 high-grade Factory presets, covering almost any imaginable application. On top of that, the M-One can store up to 100 of your favorite presets in the User bank.

## MAIN FEATURES

- 20 incredible TC effects: Reverb, Chorus, Tremolo, Pitch, Delay, Dynamics etc.
- Analog-style User-Interface
- ► 100 Factory/100 User presets
- ► Dual-Engine<sup>™</sup> design
- 24 bit A/D-D/A converters
- S/PDIF digital I/O, 44.1-48kHz
- 1/4" balanced jacks I/Os
- 24 bit internal processing

## OTHER USEFUL ALGORITHMS

Bring new life to your mixes with TC's unique Compressor and Limiter algorithms. Add incredible Delays, wide Chorus or enhance the details of your source material with the Parametric Equalizers. The M-One has even more algorithms, such as Flanger, Prtch, Gate, Expander, De-esser, Tremolo and Phaser. All the algorithms are of the well known TC Electronic sound quality. Do not waste your time programming – experiment with the many reverbs of the M-One to hear which setting suits your source material best.

## **DUAL SEND/RETURN**

Use the M-One Engines in the Dual Send/Return mode, and get two independent effects processors. Connect one Auxiliary to the left input of the M-One, and a second to the right input. The stereo output of the two Engines are now mixed internally, and can be returned to a single stereo effects return on your mixing console, giving you two full blown stereo effects simultaneously.



# THE MULTITAP RHYTHM THE MULTITAP RHYTHM



## **BELAY REINVENTED**

With the introduction of the D-Two Rhythm delay TC Electronic once again revolutionizes the concept of delay effects processing by adding the abillity to actually create rhythmic delay patterns. Based on the classic TC 2290, the Rhythm Delay feature makes the D-Two the most unique dedicated delay unit ever.

## TRUE 24 BIT

A state-of-the-art hardware platform provides you with true 24 bit resolution A/D and D/A converters and true 24 bit Ram memory. The unique hardware design ensures that the Delay Effects you get from the D-Two come in the best possible audio quality.

## RHYTHM TAP FERTURE

TC introduces the truly musical Rhythm Tap feature: Not only tempo, but actual rhythmical patterns can be tapped in directly - or quantized according to a specific tempo and subdivision. Control the exact number of repeats with Absolute Repeat Control and specify repeats of up to one second each.

## COMPLETE OVERVIEW

IC Electronic has a long tradition of building high quality intuitive user-interfaces. With the introduction of the D-Two this proud tradition is upheld. An ultra detailed Multi Spectral LCD Display provides you with complete overview of what is actually going on in your processor. Every feature has its own section in the Display. This makes it easy to take advantage of the many unique possibilities integrated in the D-Two.

## SETTING UP YOUR D -TWO

Setting up your D-Two is extremely easy. Either choose Regular mode for standard delay applications, or choose the unique D-Two Rhythm mode. Simply tap in a specific rhythm and match this to your source material. You automatically enter the rhythm mode when you press the FEEDBACK/RHYTHM key.

## SIX UNIQE DELAY EFFECTS

The D-Two Delay Processor comes with, six unique delay effects: Chorus, Filter, Spatial, Reverse, Dynamic Delay and Ping Pong. All effects within the D-Two are in the world acclaimed TC Quality. You can access all parameters of the six effects directly from the front panel.

## MAIN FEATURES

- Use the unique Rhythm Tap feature to exactly match your delay to your music
- True 24 bit RAM memory ensures the best possible audio quality
- Multi spectral LCD Display provides informative and complete overview
- The D-TWO gives you TC's world acclaimed Chorus effect
- Make a softer, warmer and more natural sounding delay with the Filter effect
- Add more width to source material by using The Spatial Effect
- Reverse Delay processes the signal and adds a backward delay
- Easy access to the wide variety of Factory presets

## THE DELAY CONCEPT

### STANDARD DELRY LINE

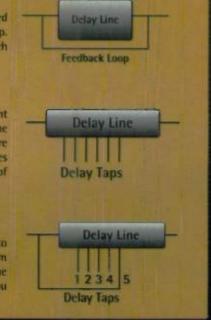
The standard delay line of the D-Two is based around a delay line with a feedback loop. The audio is fed back via the loop, which results in repeats.

### THE D-TWO DELAY LINE

The D-Two delay/feedback strategy is different from ordinary delay products. In the D-Two the repeats are caused by a number of taps that are not fed back into the delay line. This enables the D-Two to output a specific number of repeats, providing you with ultimate control.

## RHYTHM DELRY

It is also possible to feed the last tap back into the delay line, generating a complete rhythm sequence, which is repeated when you have the Feedback active. This unique feature allows you to exactly match your effects to your music.



## NOT JUST A PRETTY FACEPLATE

Get Connected: Connect your M-One or D-Two to your mixer or other studio devices in stereo or mono via the four balanced 1/4" phone jack connectors.

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Common Sense: Take your M-One or D-Two with you anywhere in the world. The internal auto-sensing power supply automatically adjusts itself to 100-240V,

## TECHNICAL SPECIFICATIONS

**DIGITAL INPUTS AND OUTPUTS** 

Connectors:

Formats:

**Output Dither:** 

Sample Rates:

Processing Delay:

Frequency Response DIO:

Connectors:

**Dynamic Range:** 

Frequency Response:

ANALOG OUTPUTS

Connectors:

Impedance Balanced /

Unbalanced:

**Output Ranges:** 

D to A Conversion:

D to A Delay:

Dynamic Range:

RCA Phono (5/PDIF)

S/PDIF (24 bit), EIAJ CP-340, IEC 958

HPF/TPDF dither 24/20/16/8 bit

44.1 kHz, 48 kHz

0.1 ms @ 48 kHz

DC to 23.9 kHz ± 0.01 dB @ 48 kHz

Impedance, Bal / Unbal:

Max. input Level:

Min, Input Level for 0 dBFS:

to D Conversion:

A to D Delay:

Crosstalk:

Max. Output Level:

1/4" phone jack balanced

21 kohm / 13 kohm

+24 dBu

@ 12 dB headroom: -12 dBu to +12 dBu

24 bit, 128 x oversampling bitstream

0.65 ms / 0.70 ms @ 48 kHz / 44.1 kHz

100 dB tvp. 20 Hz - 20 kHz

Typ < 92 dB (0,0025 %) @ 1 kHz

+0/-0.1 dB @ 48 kHz, 20 Hz to 20 kHz <-95 dB. 20 Hz to 20 kHz

1/4" phone tack, balanced

40 phm

+20 dBu (balanced) Balanced: 20/14/8/2 dBu

Unbalanced: 14/8/2 dBu

24 bit, 128 x oversampling bitstream 0.63 ms / 0.68 ms @ 48 kHz / 44.1 kHz

104 dB typ, 20 Hz to 20 kHz

Typ <-94 dB (0.002 %) @ 1 kHz.

Frequency Response: Crosstalk:

**FMC** 

Complies with:

SAFFTY

Certified to:

**ENVIRONMENT** 

**Operating Temperature:** 

Storage Temperature:

**Humidity:** 

CONTROL INTERFACE

MIDI:

Pedal:

**GENERAL** 

Finish:

Display:

Dimensions:

Weight:

Mains Voltage:

**Power Consumption:** 

WARRANTY Parts and Labor:

Note:

+0/-0.5 dB @ 48 kHz, 20 Hz to 20 kHz c-100 dB, 20 Hz to 20 kHz

EN 55103-1 and EN 55103-2

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0° C to 50° C (32° F to 122° F) -30° C to 70° C (-22° F to 167° F)

Max. 90 % non-condensing

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23 character / 280 icon STN-LCD display

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One year

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## TECH PAGE

## Quantum Mirage

ne of the biggest obstacles to the continued shrinkage of electronic elements within integrated circuits is the connection between them. As the size of these elements decreases, so must the size of the wires that carry electrons from one to another. But beyond a certain point, a wire's ability to conduct electrons is significantly hampered, preventing the message from getting through. Therefore, if nanotechnology and atomic-scale computers are to become a reality, an alternative nanotechnology. means of sending information between circuit elements must be developed.

One exciting possibility was recently announced by IBM (www.research.ibm.com). Led by Donald Eigler, a team of scientists at the company's Almaden Research Center in San Jose, California, demonstrated a remarkable phenomenon called a quantum mirage: a clearly defined "reflection" of an atom located at a different point in space. Understanding how this relates to nanocircuit communication requires some background, so bear with me.

First, we need to take a slight detour to explore a geometric shape called an ellipse, which looks like an oval.

Two important points are located on either side of an ellipse's center, on the line that divides the ellipse in half along its longer dimension. These two points are called the foci (rhymes with low sigh); each one is called a focus. Imagine a line extending from either focus to any point on the ellipse itself and another line extending from that point to the other focus-the combined length of both lines will be the same no matter where the point lies on

Eigler's team used a scanning tunneling microscope to assemble 36 cobalt atoms into an ellipse measuring a few nanometers (billionths of a meter) across. They constructed the ellipse on the surface of a single copper crystal that was IBM makes

a quantum

leap in

One more detour: cobalt atoms exhibit a property called a magnetic moment. When a cobalt atom is deposited on a metallic, nonmagnetic surface (such as copper), the electron sea produces what is called the Kondo effect, after Japanese physicist Jun Kondo, who explained the phenomenon in 1964. Basically, the

electrons near the atom align themselves to offset its

cooled to 4 degrees kelvin (that is, 4 degrees above

absolute zero) within an ultrahigh vacuum. This ellip-

tical structure, called a quantum corral, confines a portion of the two-dimensional "sea" of electrons that

exists on the crystal's surface.

magnetic moment, effectively canceling it out. The Kondo effect is highly localized and easily detected using spectroscopic techniques.

When the IBM scientists placed a single cobalt atom within the quantum corral, they saw the Kondo effect at the atom's location, as expected. But when they moved this atom to one of the ellipse's foci, something amazing happened: the Kondo effect also appeared at the other focus, even though no atom was there (see Fig. 1). The "phantom" atom is called a quantum mirage; information about the real atom is trans-

> mitted to the other focus of the ellipse via the wavelike medium of the electron sea without using any wires. Due to the wave nature of electrons, the physics of the quantum corral is analogous to the vibration of a guitar string or a drum head.

> The potential applications of this phenomenon are many and varied; for example, the presence or absence of a quantum mirage might be used to represent one bit of data in a region far smaller than any current electronic device can manage. It will be years before this technology becomes practical, but it could eventually yield computers (and the musical tools they provide) that are many orders of magnitude smaller, faster, and less power-hungry than anything we can conceive today.

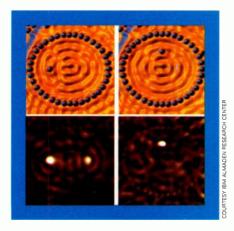


FIG. 1: When a cobalt atom is placed at one focus of the elliptical quantum corral (upper left), the Kondo effect is evident at both foci (lower left). When the atom is placed elsewhere in the ellipse (upper right), no quantum mirage appears (lower right).

Technologies converge as the century draws to a

# The Electronic Century

Part IV:

## The Seeds

of the Future

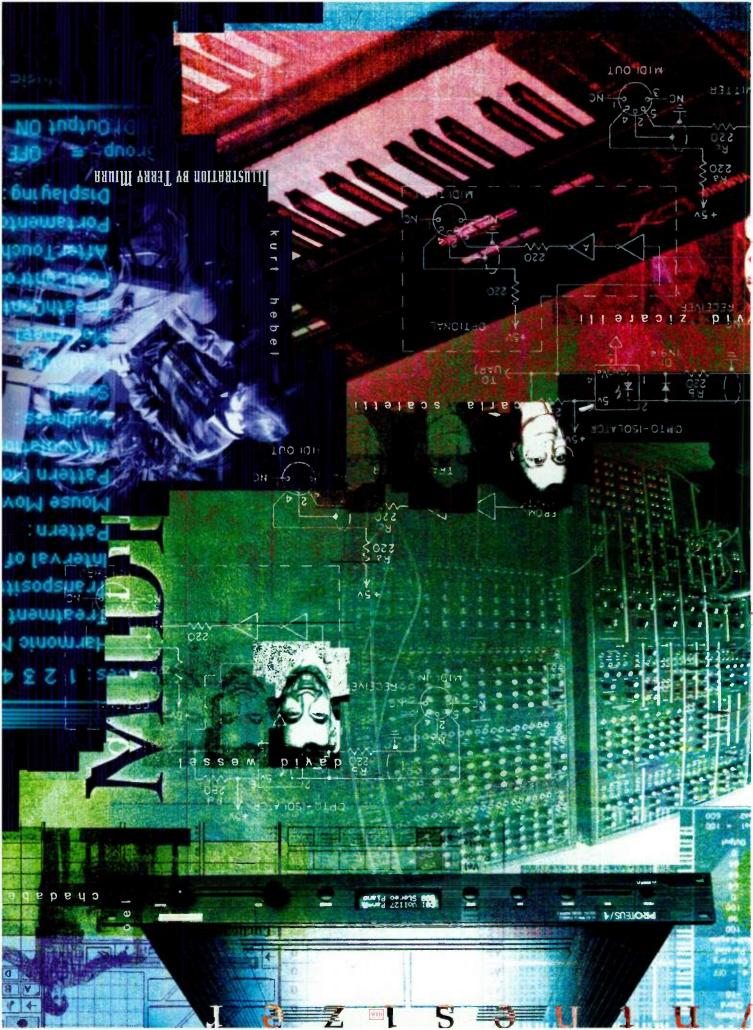
close.

Chadabe

At the end of the 1960s, two distinct but parallel paths of technical innovation traversed the field of electronic music. One of the paths, leading toward a future of digital audio and digital signal processing, was computer music. It was neither musically nor technically an easy path to follow. But the difficulties of computer-music development, such as the lack of real-time feedback and the need to specify music in computer code, were offset by the promise of creating any sound imaginable—not to mention the advantages of precise control, repeatability, and nearly indestructible storage.

The other path of technical progress, followed by many musicians, led to the development of the synthesizer. Analog synths, many of which could be played like traditional instruments, opened up a new world of electronic sound in performance. With

the help of hugely successful recordings like Wendy Carlos's *Switched-On Bach* and Keith Emerson's single "Lucky Man," synthesizers were becoming standard in virtually every band's instrumentation.



#### SYNTHESIZERS OF THE '70S

By the beginning of the 1970s, it was clear that electronic sounds were hot and that electronic music could become a viable industry. In fact, the market exploded during the decade, with many new companies developing new instruments, and the technology itself

advanced quickly. As we moved from the transistors of the '60s to the integrated circuits of the '70s, computers and analog synthesizers became less expensive and easier to use, and they were often joined together in what were called hybrid systems.

In several experimental studios-

including those at Bell Telephone Laboratories in Murray Hill, New Jersey, and the Institute of Sonology in Utrecht, the Netherlands—computers were used as sophisticated sequencers to generate control voltages for analog synthesizers. Emmanuel Ghent's *Phosphones* (1971) and Laurie Spiegel's *Appalachian Grove* (1974) are examples of music created at Bell Labs; Gottfried Michael Koenig's *Output* (1979) exemplifies music composed at the Institute of Sonology (see the sidebar "Recommended Resources").

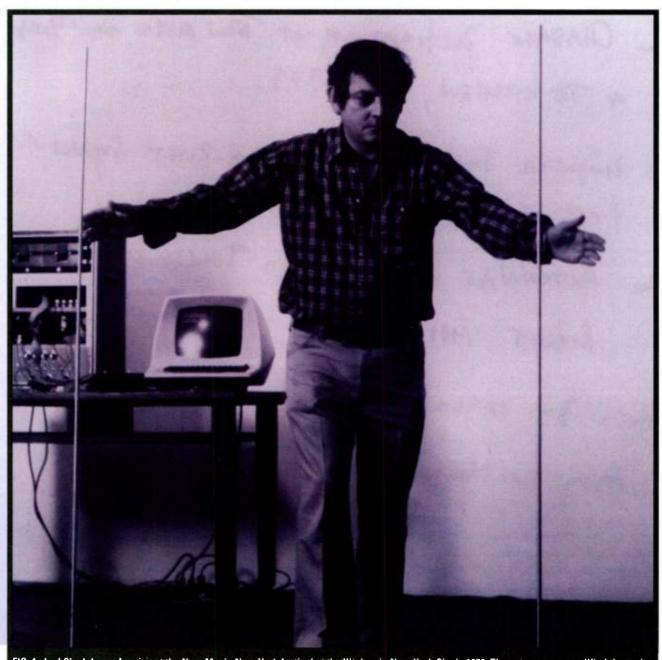
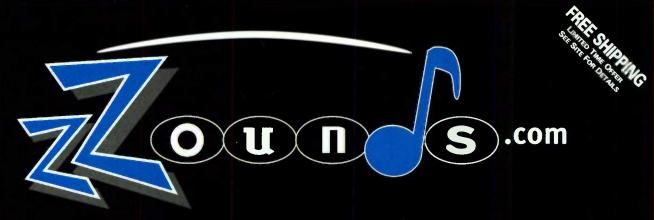


FIG. 1: Joel Chadabe performing at the New Music New York festival at the Kitchen in New York City in 1979. The antennas are modified theremins that were custom-made by Robert Moog. Here Chadabe is using them to "conduct" the first Synclavier, which is on the table behind him.

PHOTO BY CARLO CARNEVALI/COURTESY ELECTRONIC MUSIC FOUNDATION

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The most important trend of the '70s, however, was the increasing accessibility of digital technology. With the invention of digital synths, the analog and digital paths—which had wound their separate ways through the landscape of electronic music in the '60s—began to converge. These new instruments

combined the performance capabilities of analog synthesizers with the precision of computers.

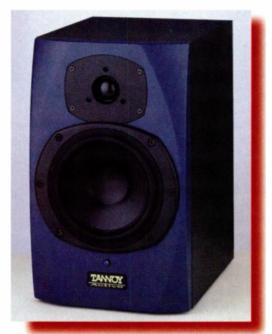
In 1972 Jon Appleton was director of the Bregman Studio at Dartmouth College, which housed a large Moog modular system. Appleton asked Sydney Alonso, a faculty member at Dart-

mouth's Thayer School of Engineering, about using a computer to control this system. Alonso's advice was to forget the Moog and build a digital synthesizer. Together they did, calling it the Dartmouth Digital Synthesizer. Cameron Jones, a student at the college, wrote the software. Alonso and Jones then formed a company called New England Digital and, with Appleton's musical advice, went on to create the Synclavier.

The Synclavier was a computer-anddigital-synthesizer system with an elegantly designed keyboard and control panel. In September 1977, I bought the first Synclavier, although mine came without the special keyboard and control panel that Alonso and Jones had so painstakingly designed (see Fig. 1). My idea was to write my own software and control the computer in various wavs with a number of different devices. For example, in Follow Me Softly (1984) I used the computer keyboard to control the Synclavier in a structured improvisation with percussionist Jan Williams. In 1983, Appleton composed Brush Canyon for a Synclavier with both the keyboard and the control panel.

By the late '70s, digital synthesizers were under development at research institutions such as Bell Labs and the Paris-based organizations Groupe de Recherches Musicales and Institute for Research and Coordination of Acoustics and Music (IRCAM). The market was full of analog, hybrid, and all-digital synthesizers, drum machines, and related devices. These products were manufactured by a long list of companies, among them ARP, Crumar, E-mu Systems, Kawai, Korg, Moog Music, Oberheim Electronics, PPG, Rhodes, Roland, Sequential Circuits, Simmons, Synton, and Yamaha. Technology was advancing quickly, the level of creativity was high, a new mass market was emerging, and price was increasingly important. High-end products were quickly adapted to a larger market. When Fairlight Instruments put the first sampler on the market in 1979, it cost about \$25,000; by 1981, E-mu's Emulator was selling for \$10,000. It was an exciting time, with new and powerful technologies appearing at increasingly affordable prices.

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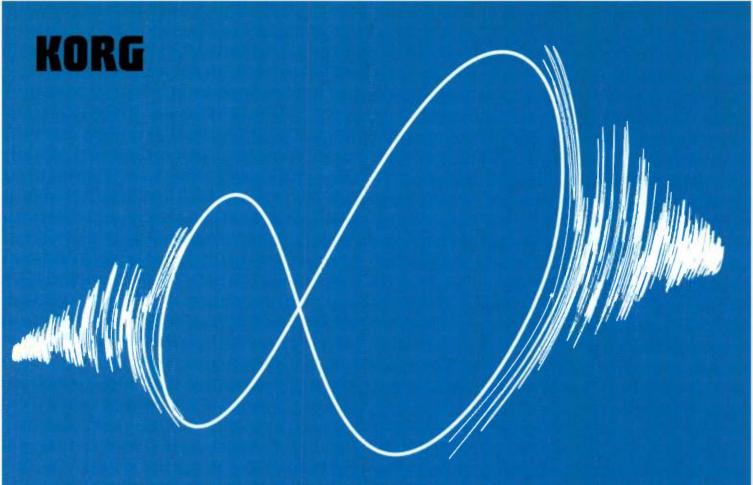
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### THE BEGINNING OF MIDI

Although innovation, creativity, and adventure were in the air at the end of the '70s, there was also a large measure



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# The Seeds of the Future

of chaos in the market. Standardization was nonexistent: if you bought a synthesizer from one manufacturer, you had to buy other products from that same company to maintain compatibility. The marketplace was fragmented, with no fragment large enough to warrant major investment. In the view of Roland president Ikutaro Kakehashi, standardization was necessary to make the industry grow. With a global market unified by a digital standard, a company of any size could develop and sell its products successfully.

In June 1981, Kakehashi proposed the idea of standardization to Tom Oberheim, founder of Oberheim Electronics. Oberheim then talked it over with Dave Smith, president of Sequential Circuits, which manufactured the extremely successful Prophet-5 synthesizer. That October, Kakehashi, Oberheim, Smith, and representatives from Yamaha, Korg, and Kawai met to discuss the idea in general terms.

In a paper presented in November 1981 at the AES show in New York, Smith proposed the idea of a digital standard. At the NAMM show in January 1982, Kakehashi, Oberheim, and Smith called a meeting that was attended by representatives from several manufacturers. The Japanese companies, along with Sequential Circuits, were the primary forces behind sustaining interest in the project, and in 1982 they defined the first technical

specification of what came to be known as the Musical Instrument Digital Interface, or MIDI. At the January 1983 NAMM show, a Roland JP-6 was connected to a Sequential Circuits Prophet 600 to demonstrate the new MIDI spec. After some refinement, MIDI 1.0 was released in August 1983.

The adoption of MIDI was driven primarily by commercial interests, which meant that the specification had to represent instrumental concepts familiar to the mass market. Because that market was most comfortable with keyboards, MIDI was basically a spec designed to turn sounds on and off by pressing keys. For some musicians, this was a serious limitation, but most felt that the benefits of MIDI far outweighed its shortcomings.

## FROM FM TO SAMPLES

In business terms, MIDI was a smashing success. Its universal format allowed any company—new or established, large or small—to present the world with an original concept of music.

In 1983, Yamaha introduced the first monstrously successful MIDI synthesizer. The DX7 was a hit not only because of its MIDI implementation, but also because it sounded great and was reasonably priced at less than \$2,000. To generate sounds, the DX7 used frequency modulation (FM), which John Chowning had developed at Stanford University in 1971 and



FIG. 2: Among the earliest digital samplers was the Ensoniq Mirage. It was supported by numerous third-party manufacturers that offered both hardware accessories and software enhancements.

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# The Seeds of the Future

which Yamaha had licensed in 1974.

FM results when the amplitude of one waveform, called the modulator, is used to modulate the frequency of another waveform, called the carrier. As the amplitude of the modulator increases, the spectrum of the carrier spreads out to include more partials. And as the frequency of the modulator changes, the frequencies of the partials in the carrier spectrum change. In other words, by changing the amplitude or frequency of the modulator, a performer can change the spectrum's bandwidth and the timbre of sounds. The early advantage of FM synthesis was that simple controls could cause major changes, making instruments like the DX7 very popular for live performance.

Throughout the '80s, Yamaha continued to develop new applications of FM synthesis in a line of instruments, while many other companies—Akai, Korg, and Roland among them—developed their own synthesizers. Roland, for example, released the Juno-106 in 1984 and the D-50 family in 1987. To a growing number of musicians, however, the main disadvantage of synthesized music was that it sounded

electronic. As it turned out, most MIDI musicians wanted emulative sounds. They turned to samplers, which allowed any sound—whether trumpet riff or traffic noise—to be recorded and played back at the touch of a key.

In the early '80s, E-mu Systems had broken through the first major price barrier in the sampler market with its \$10,000 Emulator. In 1984, Ensoniq introduced the Mirage at less than \$1,300 (see Fig. 2). And in 1989, E-mu lowered the bar even further. Its Proteus, a sample-playback device that came with 256 prerecorded samples and an exceptionally simple interface, cost less than \$1,000.

The electronic-music industry continued to grow throughout the 1980s. By the early '90s, the market was overflowing with synthesizers, samplers, and other MIDI hardware, but attention was beginning to center on software development.

## **SOFTWARE BEGINNINGS**

A MIDI software industry had already emerged in the mid-'80s. For example, Opcode Systems established itself in 1984 with a MIDI sequencer for the Macintosh and almost immediately

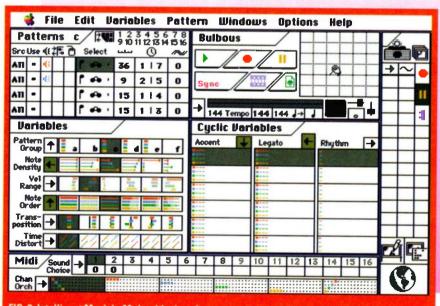


FIG. 3: Intelligent Music's *M* algorithmic software was a remarkable tool for generating music on a Macintosh. The software also had a brief life span on the PC and has recently been reintroduced for the Mac by Cycling '74, a company founded by *M*'s inventor, David Zicarelli.



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# The Seeds of the Future

expanded its product line to include David Zicarelli's DX7 patch editor. At the same time, other companies were forming and releasing similar software, among them Steinberg Research in Hamburg, Germany, and Mark of the Unicorn in Cambridge, Massachusetts.

As personal computers got faster and less expensive and computerbased MIDI sequencers became more commonplace, other MIDI software applications were developed. In 1985, Laurie Spiegel wrote Music Mouse, a program that contained harmony-generating algorithms. In 1986, Zicarelli developed two applications, M and Jam Factory, for Intelligent Music, which continued to develop other interactive-composing programs during the years that followed. Of particular interest was M, an interface of musical icons that controlled algorithms (see Fig. 3). Given a melody or other input, a composer could use M to generate an infinite

stream of rhythmic and melodic variations, ending with something distinct and original. For example, I composed After Some Songs, a group of short improvisational compositions for electronics and percussion, by using M to transform some favorite jazz standards.

In 1985, Miller Puckette went to IRCAM to develop software for the 4X, a digital synthesizer built by Giuseppe di Giugno. By mid-1988, Puckette had developed a graphical modular control language that he called *Max*. At about the same time, Zicarelli saw a demonstration of *Max* and, after discussing it with Puckette, developed the language as a commercial product—



FIG. 4: Pictured (from left to right) are Bill Walker of the CERL Sound Group, Kurt Hebel, and Carla Scaletti with the Platypus at a sound check before a November 1989 concert in Columbus, Ohio.

PHOTO BY LIPPOLD HAKEN/COURTESY SYMBOLIC SOUND



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# The Seeds of the Future

first with Intelligent Music and then, after Intelligent changed directions in 1990, with Opcode Systems. *Max* was released in late 1990 and remains an essential tool for many customized music software applications.

The first digital audio programs were also developed in the mid-'80s. Among them were MacMix, written by Adrian Freed in 1985, and Sound Designer, a Digidesign product released the same year. In 1986, working in conjunction with a company called Integrated Media Systems, Freed added a specialized hardware device and called the system Dyaxis. And in 1988, taking advantage of increasing computer speeds, larger hard drives, and digital-to-analog converters, Digidesign released Sound Tools, which established an industry standard in audio editing. Digital audio was fast becoming accessible to musicians.

### TRENDS INTO THE '90S

Everything expanded throughout the 1990s. The market filled with increas-

ingly sophisticated synthesizers, samplers, drum machines, effects generators, and an enormous variety of modules, each of them doing something different and offering unique sonic possibilities. A large secondary market of patch panels and other MIDI-management gear formed around the needs of professionals with racks of devices. Software applications, including sequencers, patch editors, effects processors, and hard disk recording systems, also permeated the market. In fact, there was so much to learn that the following joke was frequently heard: "What's a band?" "Three guys reading a manual."

Some musicians may have felt a few pangs of nostalgia for analog equipment and sounds, but the 1990s were largely a digital decade driven by the availability of ever-faster microprocessors. Not surprisingly, as personal computers kept getting speedier and more powerful, digital audio became increasingly software based.

But the advances of the 1990s reach far beyond simply editing sound. Digital signal processing (DSP), which allows composers to transform as well as synthesize sounds, became an important component in digital audio systems. One of the most complete DSP applications to appear in the late '90s was MSP, created by Miller Puckette and David Zicarelli and marketed through Cycling '74, a company that Zicarelli formed in 1997 to develop DSP software as well as to make M available again.

Among the pioneers in DSP systems for composers is Carla Scaletti. In 1986, she began creating a software synthesis system that she called Kyma (Greek for wave). By the following year she had extended Kyma to include the Platypus, a hardware audio accelerator built by Kurt Hebel and Lippold Haken that sat alongside a Macintosh, received instructions, and generated sound (see Fig. 4). Scaletti's 1987 composition sunSurgeAutomata demonstrates the sound-processing and algorithmic abilities of the Platypus. By 1990, Scaletti and Hebel had upgraded the hardware to a system called the Capybara. In 1991, they formed Symbolic Sound Corporation and shipped the first complete Kyma system, available initially for the Mac and shortly thereafter for the PC. With its evolving hardware and continual upgrades, Kyma remains one of the most powerful sound-design systems available today.

## RECOMMENDED RESOURCES

For an overview of electronic-music history, read *Electric Sound*, by Joel Chadabe (Prentice Hall, 1996).

For an overview of MIDI, read MIDI for the Professional, by Paul D. Lehrman and Tim Tully (Amsco Publications, 1993).

The following compact discs feature music mentioned in this article:

After Some Songs (Deep Listening) is a group of Joel Chadabe's abstractions of jazz standards, for computer and percussion.

CDCM Computer Music Series, volume 3 (Centaur), includes Carla Scaletti's sunSurge-Automata, for the Platypus.

CDCM Computer Music Series, volume 6 (Centaur), includes Jon Appleton's Brush Canyon, for Synclavier.

CDCM Computer Music Series, volume 24 (Centaur), includes Joel Chadabe's Follow Me Softly, for Synclavier and percussion, and Cort Lippe's Music for Clarinet and ISPW.

Computer Music Currents 2 (Wergo) includes Emmanuel Ghent's Phosphones, composed at Bell Telephone Laboratories in 1971.

Gottfried Michael Koenig (BVHaast) includes Koenig's Output, composed in 1979 at the Institute of Sonology.

Women in Electronic Music 1977 (CRI) includes Laurie Spiegel's Appalachian Grove, composed at Bell Labs in 1974.

These and other interesting items are available from CDeMusic at www.cdemusic.org.

#### INTO THE 21ST CENTURY

As we look to the future, it's hard to know which innovations will have the greatest impact on our lives and our work. Although we can assume that digital audio technology will keep improving as computing horsepower increases and prices drop, predicting exactly how this will play out in our studios isn't easy. I asked several leading figures in the music-technology field for their thoughts on what the next decades will bring. Here are some of their predictions:

Craig Harris (composer, author, and executive editor of the Leonardo Electronic Almanac): "New instruments will have enormous flexibility in both the sonic realm and the modes of interaction, such that composers can create in the way that is most effective for them, performers can realize works in ways that work best for their own personal



# The Seeds of the Future

styles, and audience members can benefit from a rich variety of interpretations. This is one realm that distinguishes electronic instruments from traditional instruments, in that there is no preconceived sonic realm or method for interaction that is inherent in the machine. For the first time, we have instruments that will have their limits established more by our imaginations than by the laws

tions than by the laws of acoustics."

Carla Scaletti (composer, software developer, and president of Symbolic Sound Corporation): "What seems to interest us is the process of making music. Bootlegs of tour improvisations on original album material are more sought after than the finished albums. Some musicians are beginning to post MP3 versions of 'works in progress' on the Internet. so all of us can witness

and participate in the process of exploration and refinement that goes into a 'finished' album. Every album that is released immediately spawns multiple offspring in the form of remixes. Interactive and immersive environments like computer games require music that can be 'traversed' in a multiplicity of ways; each path through the game results in a new piece of music. The 21st century will be 'the compo-

sition century' where 'objects' (like finished albums) will be virtually free on the Internet, while the creators of those objects will be highly sought after."

Daniel Teruggi (composer and director of the Groupe de Recherches Musicales in Paris): "If we put our analytical ears on, we see that there is still a great difference between a recorded sound and the sound produced and propagated by an acoustical device.

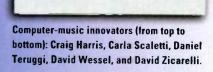
Loudspeakers, microphones, amplifying systems, and digital conversion are the elements of sound processing that still have to achieve what I would call a more 'realistic' image of sound."

David Wessel (researcher and director of the Center for New Music and Audio Technologies at the University of California, Berkeley): "Standard generalpurpose processors like the PowerPC and Pentium are now capable of realtime music synthesis and processing. Laptops will become the signal processors and synthesis engines of choice, at the core of the new performanceoriented electronic music instrumentation. I'm confident that we will also see the development of a new generation of gesture-sensing systems designed with music in mind, including a number of the common interfaces like drawing tablets and game controllers adapted for the intimate and expressive control of musical material. And I see, and am beginning to hear, the emergence of an electronic music that might be more akin to chamber music or that of a small jazz group where musical dialog and improvisation play essential roles.

David Zicarelli (software developer and president of Cycling '74): "The computer, synthesizer, and tape recorder have become the new folk instruments of industrialized cultures, replacing the guitar. An overwhelming number of recordings are being produced in the electronica genre right now, and there is no sign that this will stop anytime soon."

You may now be wondering how to take advantage of the resources available to you right away and in the years to come. Start by thinking about what kind of music you want to write and which tools will best help you reach your goals. Explore the Web and read magazines such as EM for new developments in hardware and software. Above all, learn the history: read books on the subject and study recordings by the pioneers as well as the current movers and shakers. A 100-year tradition is waiting to be explored, and the more you know about the past, the better you can shape your own future.

Joel Chadabe is a composer, past president of Intelligent Music, author of Electric Sound, and president of the Electronic Music Foundation. He can be reached at chadabe@emf.org.



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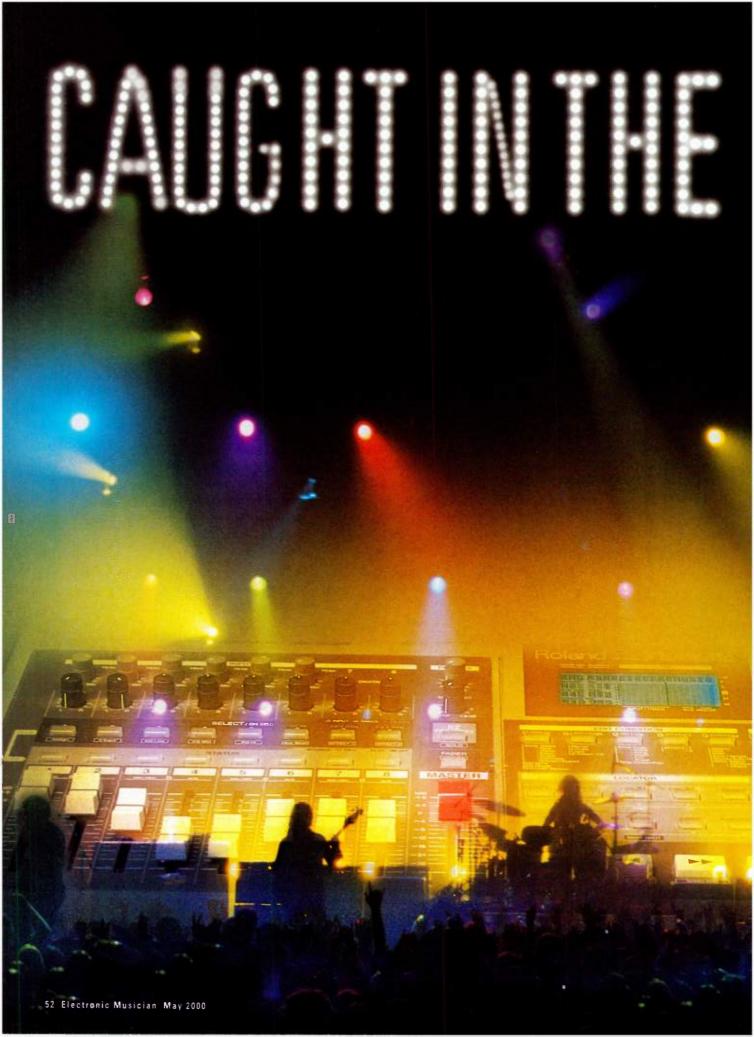
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## **RECORDING LIVE**

## CAN BE A CHALLENGE,

## BUT THE REWARDS ARE GREAT.

## By Mike Levine

ive recording offers musicians the opportunity to capture the unique energy and ensemble interaction of a performance. It's especially beneficial for groups that excel on stage but can't seem to re-create the same musical magic in the studio. As almost any gigging musician will tell you, there's something about playing for an audience—especially a receptive one—that brings out the best in a band.

From an engineering standpoint, though, recording live is akin to working without a net. There's little margin for error, and unlike the controlled environs of a recording studio, the working conditions are often unpredictable.

Typically, an engineer recording a live show works in the midst of crowds, smoke, and excessive sound levels. With the exception of those occasions when the gear is set up in an isolated room or in a truck outside the venue, monitoring must be done with headphones in the room where the band is playing. Through all of this, the recordist must valiantly fight to maintain acceptable recording levels while keeping the twin demons of noise and distortion at bay.

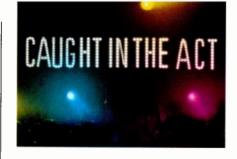
Engineers on live projects also have to polish their diplomacy. Often they'll need to negotiate with the house sound engineer for permission to set up extra mics, install splitters by the stage box, or tap into the front-of-house (FOH) console.

These challenges notwithstanding, a live recording can yield quality tracks for a CD, MP3, or video at a fraction of the cost typically incurred in a commercial studio. Moreover, if the show is being recorded direct to 2-track, no mixing is required afterward—the project can proceed directly to the mastering stage, reducing expenses even further.

If you are contemplating doing a live recording, whether as a performer, an engineer, or both, you have many options to consider. I solicited the viewpoints of a number of live-recording engineers and found that although they agree on some issues—such as the importance of preparation—there is a great diversity of opinion regarding gear, formats, and methodology. Ultimately, the best approach depends on a number of factors, including the budget, the purpose of the project, and the type of equipment that you own or have access to.

The most fundamental decision is whether to record in stereo or to multitrack. The choice you make is likely to have a significant impact on the complexity and expense of the project as well as on the sound of the finished master. I'll start with the simpler applications and then proceed to the more complex.

WRH



#### **A STEREO PAIR**

Generally speaking, stereo recording is the easiest and least-expensive way to capture a live performance—especially when using an affordable format such as DAT or MiniDisc. (Of the two. DAT is preferable because it has better sound quality and allows you to record for up to two hours without changing tapes.) All you need is the recorder, a couple of cables, a microphone stand or two, and a pair of identical mics (or a dedicated stereo mic). After setting levels, simply press Record and let it roll. Assuming you use appropriate, good-quality mics and position them well, you should end up with a recording that offers the listener a reasonable representation of what the music sounded like in the room that night, replete with audience members talking, glasses clinking, and other reallife sounds.

Depending on the distance of the mics from the P.A., you'll pick up varying degrees of room sound. In a venue with great sound, this can be advantageous; in a space with less-than-desirable acoustics, it can work against you.

Another problem inherent to stereo 2-track recording is that success depends on the house engineer doing a good job on the mix. If the mix in the room is substandard, the recording will be substandard, too.

### **OFF THE BOARD**

Another method for making a 2-track live recording is to take a feed from the stereo bus of the FOH console directly into the recorder. If the goal is simply to make a rough document of the performance, this should suffice. However, producing master-quality recordings this way is extremely difficult.

The main problem with board tapes results from the fact that the sound engineer's objective is to get a good sound in the house. Typically, this means bringing into the mix only those instruments that need reinforcing in the room (for example, vocals, keyboards, and kick drum) and omitting those that are loud enough on their

own (for example, electric guitar, bass guitar, and the rest of the drum kit). In such a case, a direct feed from the console's stereo outputs will produce a poorly balanced recording, with some instruments well represented and others barely represented at all. "You're really at cross-purposes with the people running the sound system," says Philip Perkins, a production sound mixer for film and television who does a great deal of location recording, "especially in smaller spaces."

The best way to record off the board to a 2-track recorder is to set up an independent mix. This can be done a number of ways, depending on the console. The preferred way—if the board allows for it—is to employ a secondary mix bus (such as Mix B on the Mackie 8-Bus series), which provides separate volume, pan, and even rudimentary tone controls for each channel.

Another option is to feed the 2-track recorder from two spare auxes. However, auxes don't provide pan controls, so you have to apply the auxes unevenly to pan the instruments across the stereo field. Panning hard left and right is easy—you simply send aux 1 to the left input and aux 2 to the right. However, if you want to position an instrument at, say, 9 o'clock, you need to dial in a portion of the signal from aux 2, but not as much as is coming through aux 1. This is an imprecise way to operate.

#### **NOW HEAR THIS**

One of the biggest challenges in setting up an independent mix off the board is hearing things accurately. No matter how well positioned the house engineer is for setting up the P.A. mix, mixing for tape is hampered by the fact that there is insufficient isolation between the stage and the P.A. sound. This makes the process of judging levels very tricky.

Monitoring through headphones can help, but because of all the sound in the room, you're still unlikely to hear things well. If you must use headphones, select closed-ear circumaural models that provide as much isolation as possible (see Fig. 1). Some engineers build their own custom headphones by mounting high-quality drivers inside airport hearing protectors. But even with well-isolated cans, monitoring through headphones in a room awash

with loud music tends to be difficult. It's especially hard to judge the low end and to detect unwanted noise on individual tracks.

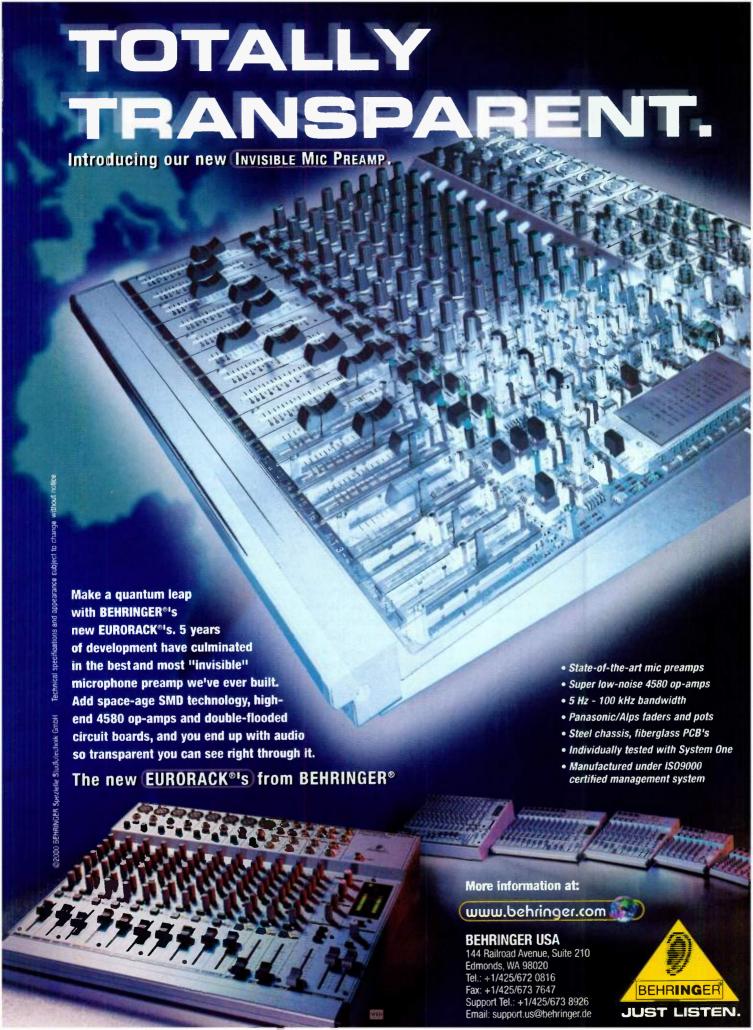
For these and other reasons, most high-quality, direct—to—2-track live mixes are done from a remote location (an isolated room or a recording truck). In a remote setup, the signals are usually split and sent to a separate console through an audio snake (see the section "Splitsville"). One major advantage of remote recording is the ability to listen through studio monitors, which is superior to mixing with headphones.

When recording direct to 2-track—and especially when using headphones as your only monitor source—you should record the sound check, then listen back and make necessary adjustments before the show starts. Likewise, if there's a break during the show, take that opportunity to listen to your mix and check for problems.

No matter how you send the signals to tape, board recordings typically suffer from a "flat" sonic quality due to insufficient room and audience sound. If the situation allows, set up a stereo pair of room mics (or a stereo mic) and mix the ambient sound with the other sources from the board. This requires careful blending, but it definitely livens up the sound. When done well, it can yield excellent results (see the section "A Room with Two Mics").



FIG. 1: Not every headphone provides sufficient isolation for monitoring live recordings. Look for models with a circumaural earcup design—that is, earcups that completely encircle and enclose the ears—such as the Sony MDR-7509.





### **MULTI DOES IT**

If your aim is to make a CD or even a broadcast-quality recording from a live show, consider recording live-to-multitrack. Without exception, the audio engineers I spoke to agreed that this is the best way to get premium-quality results.

The obvious advantage to using a multitrack for live recording is that it offers greater flexibility when you're shaping the final sound. Yes, mixdown is required, which means more work and greater expense; however, the control over the end product often justifies the additional expenditures. Not only can you set levels, add effects, and equalize tracks after the fact, but you can also "cheat" if necessary by replacing off-key or lame parts or even by overdubbing extra tracks to supplement the mix. Although some people may object to them on aesthetic grounds, after-thefact overdubs are quite common in professional live recording—and they may allow you to salvage songs that otherwise would be unusable.

MDM. Of the numerous multitrack formats available, perhaps the one that most readily lends itself to live recording is the modular digital multitrack (MDM). This format is compact and rack-mountable, and it allows you to record for a substantial period without switching tapes. The Tascam DTRS format provides the longest recording times, offering up to 1 hour and 53 minutes on a single tape. The latest Alesis ADATs can record only up to 67 minutes continuously (which is more than sufficient for most club sets); however, they offer the advantage of 20-bit recording, as compared with the Tascam DA-38's and DA-88's 16-bit resolution. Then again, if you're ready for 24-bit recording, check out Tascam's DA-78HR and the soon-to-be-released DA-98HR.

Another good reason to use MDMs is that you can easily gang multiple machines together to add more tracks. This is important: to produce a high-quality recording of a four- or five-piece band, you're probably going to need at least 16 tracks, if not 24.

DAW. While many engineers prefer MDMs, others happily use hard disk systems—both computer-based and stand-alone—for live recording. Recording to hard disk offers many advantages, but it comes with its own set of problems.

Sonically, a computer-based digital audio workstation (DAW) is hard to beat. Many systems feature 24-bit recording, providing noticeably better fidelity and headroom than 16- and 20-bit systems. According to Joel Singer, marketing manager for Audio-Technica (as well as an engineer with years of live-recording experience), "Going from 16- to 24-bit is a huge jump. The depth of the recording is much greater in 24-bit, and everything just sounds more three-dimensional."

DAWs also offer a virtually unlimited track count—that is, assuming the CPU and hard disk are fast enough. This can be extremely advantageous because more elements can be tracked individually, which in turn provides more flexibility at mixdown.

Taking your computer to the venue has several drawbacks. For one thing, desktop computers are designed to be stationary, not portable. They're also relatively delicate. If you decide to use a computer for location recording on a consistent basis, you'd be wise to get it rack-mounted. (See "DIY: Rack-Mounting Your PC" in the March 2000 issue of EM.) Computer-based systems are also more vulnerable to noise emanating from the stage's electrical circuits (which often run the house lights)—usually a major source of interference.

Another disadvantage of recording to a computer-based DAW is that it's not as stable as tracking to tape. No matter how loaded and up-to-date the computer system is, it can still crash—possibly in the middle of a song.

HDR. One viable option is to use stand-alone hard disk recorders (HDRs), including modular models (for example, the Akai DR16) and portable digital studios (such as the Roland VS-1680). Many of these machines offer 24-bit recording, the ability to lock multiple machines to-

gether to increase track count, and a built-in mixer—something no MDM can match.

Although HDRs are generally more stable than computer-based DAWs, both contain hard disks, and sooner or later every hard disk will crash. Therefore, if you choose to go this route, back up your data immediately after the show, if not after each set.

#### **SPLITSVILLE**

Whether you're recording to 2-track or multitrack, getting the signals into the recorder cleanly is the key to making a good live recording. One of the best ways to do this is to use a splitter (see Fig. 2). A splitter acts like a glorified Y-cable, duplicating the outputs of the stage mics and direct boxes and providing the recordist with a separate feed for each sound source (see Fig. 3). Many types of splitters are available, including rack-mountable units, cable splitters, and splitter snakes. In its simplest form, a splitter is a box consisting of one female XLR input and two male XLR outputs per channel. Using splitters that contain isolation transformers is highly recommended, as these help prevent the occurrence of ground loops—a major bugaboo in live-recording situations.

In a best-case scenario, you would probably set up some additional instrument and room mics to supplement those being used in the house sound system. Ideally, the outputs of the split and supplemental sources would be sent via snake to a remote truck or isolated room. There they



FIG. 2: Though not inexpensive, splitters give recordists a great deal of control over the signal path by allowing them to bypass the house console and use the mic preamps of their choice. This Rapco 20-Channel Splitter Snake is available with and without transformer isolation for \$1,527 and \$917, respectively.

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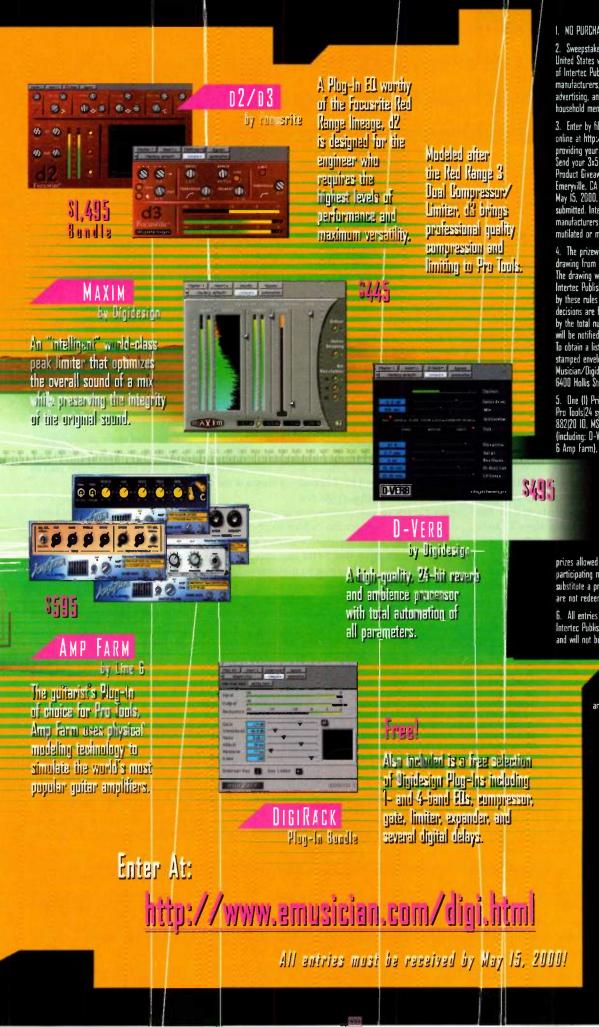
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In a multitrack situation, room mics are useful for more than simply picking up audience applause at the end of songs. The mics can also be used to capture room ambience and the overall sound of the performance. These signals, in turn, can be added during mixdown to increase "liveness" and the sense of "being there."

Probably the most common way to mic a room is with a stereo pair of cardioid condensers in an XY configuration (see Fig. 5a). The microphones are positioned on a stand (a stereo bar is helpful for this application) near the mix position and elevated seven or more feet in the air so as to reduce the pickup of audience noise and reflections from the floor.

Some engineers, like Joel Singer, prefer to use more than two mics. "In addition to an XY pair positioned to capture the sound of the P.A. in the room," explains Singer, "I also like to put up two stage mics: a stage left and a stage right. Generally, these are shotgun-type microphones—especially in larger venues—which let me really home in on the audience."

Not everyone thinks XY is the way to go. Philip Perkins, for example, prefers ORTF, a near-coincident technique developed in France for recording classical music. (This technique was also often used-from the FOH position-by "tapers" at Grateful Dead shows.) In ORTF, a pair of cardioid mics is angled 110 degrees to either side of center stage with the capsules roughly seven inches apart (see Fig. 5b). "To my ears," says Perkins, "XY miking is just not very interesting." However, he cautions that ORTF doesn't sum to mono as well as XY does.

Another tried-and-true approach employs a spaced pair of omnidirectional mics. Engineer Rich Tozzoli, who has recorded live projects for

artists such as Al DiMeola and the Beach Boys, calls this method *split omni*. Tozzoli uses a pair of Earthworks TC-30K omnis, placed separately on either side of the room directly facing the P.A. stacks and as far back as the mix position. The stereo picture captured by these mics provides the basis for Tozzoli's recordings, which he then supplements with feeds from individual instruments taken from the board or

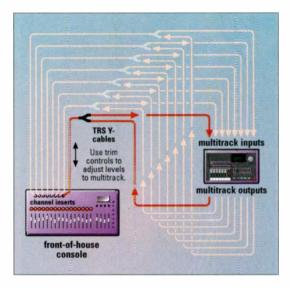


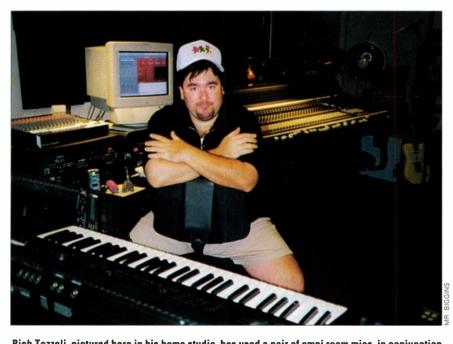
FIG. 4: One way to tap signals off the front-of-house console is to plug Y-cables into the channel inserts, sending the signals to the multitrack inputs and returning them from the multitrack outputs. Input levels to the recorder can then be adjusted with the mixer's trim controls.

from splitters. "The trick is to have the omnis do most of the work," explains Tozzoli. "I use the other mics just to tighten up the mix. Basically, if your goal is to put the record out using the two tracks from the omni mics alone, then the rest is gravy."

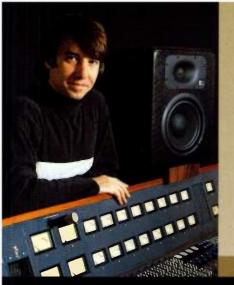
If you plan to incorporate stereomiking into your live recordings, remember that room acoustics will play a large role in the success of your productions. The old "garbage in, garbage out" axiom applies here: if the room sounds good, miking it well can yield excellent results; if the room doesn't sound good, it's probably not worth the trouble.

Therefore, Tozzoli advises bands to be as selective as possible when choosing a venue for a live recording. "Bands need to think this through ahead of time," he says. "For example, if they're in a city and they have a choice of four venues, they should go out and listen to all four. Stand in the audience and listen to what the room sounds like during a performance. Then pick the venue that sounds the best and make your recording there." Tozzoli recommends going for the "tightest sounding" room. He also finds that rooms with the most wood (floors, walls, and so on) tend to yield the best recordings.

Of course, even in a room with great sound, mic placement is critical. As mentioned previously, positioning the mics near the mix position is a good



Rich Tozzoli, pictured here in his home studio, has used a pair of omni room mics, in conjunction with feeds from the stage, to record live shows for artists such as Al DiMeola and the Beach Boys.



William Wittman is a multi-platinum Producer/Engineer, former Staff Producer/A&R Vice President (RCA/BMG Records and Columbia/Sony Records), Musician and Songwriter. His career truly covers all the bases.

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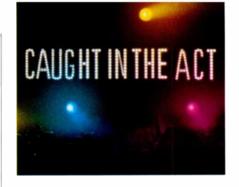
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rule of thumb. But be aware of any nearby, unwanted sounds that may get picked up. For example, if the cocktail bar is just to the right of the mix position and you have positioned your XY pair of mics nearby, one mic is going to capture an abundance of clinking glass, bar chatter, and cashregister noise—not something you want in your mix.

Obviously, a competent sound person mixing the house is indispensable, because your stereo room mics will be capturing that mix in the room. If the house sound is bad, your results will be, too.

Interestingly, some engineers avoid stereo-miking altogether—that is, for capturing the band's performance. For example, Brian Kingman, who runs Six String Recording (a successful liverecording company in New York City), uses an array of audience mics to pick up applause and such. This includes two mics at the front of the stage, pointing out toward the audience, and four more arranged around the mix position. "I prefer to get all of the band sound from the close mics alone," Kingman explains. "The ambient mics that I put up are strictly for capturing audience response."

## FROM THE SOURCE

For the most part, the live-recording engineer is dependent on the mics used in the house sound system. However, there may be occasions when the house isn't miking everything to the

degree required for the recording. In that case, you're best off requesting permission from the sound person to put up some extra mics where necessary.

For instance, Kingman finds that club sound engineers typically don't put enough mics on the drums, so he puts up more whenever possible. "I like to cover all of my bases," says Kingman, "so I mic everything I can, including the individ-

ual toms. Sometimes I even use two mics on the kick—a Shure 91 to capture the attack of the beater and a Sennheiser E602 for the low thump."

Even if the house has put up enough mics, some may not be of particularly good quality. If possible, substitute better mics, especially for picking up vocals. To minimize leakage, it's helpful to use mics with good off-axis rejection. For vocals, a number of engineers suggested using the Shure Beta 58A. The Audix OM-5 and Audio-Technica 4054 were also highly recommended.

Direct boxes used in clubs may also be substandard. Therefore, it's advisable to carry your own high-quality DIs.

Capturing a consistent signal from moving sources is a big challenge. Sax players, for example, typically move about quite a bit on stage. While this may not prove problematic for the house mix, it can cause troublesome level fluctuations on the recording. The obvious solution is to use a quality clip-on mic.

Acoustic guitar can also pose problems due to player movement. Although live-sound engineers are often content to take a line signal from an acousticguitar pickup (routed through a direct box), this may not suffice if you want to capture a more authentic guitar sound. Again, a clip-on mic (preferably a condenser) is the solution. Of course, attaching it can be problematic. One trick is to tape a tongue depressor to the side of the guitar so that one end protrudes past the guitar in the direction of the headstock (see Fig. 6). Be sure to use tape that leaves little or no residue once it's removed (for example, Permacel gaffer's tape). After attaching the depressor to the guitar, clip the mic onto the depressor and aim it at the 12th fret. No matter which way the guitarist moves, the mic signal remains constant.

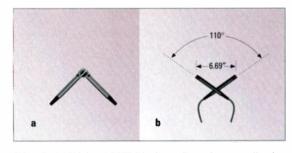


FIG. 5: Both XY (a) and ORTF (b) mic configurations are effective for stereo recording live—to—2-track. The ORTF setup creates a wider image and more sense of depth but typically produces comb-filtering effects when summed to mono.

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WRH



Some instruments are best recorded using both a mic and a direct signal, with each feed going to its own track. Electric bass is a good example. Although many engineers will simply use a DI, it's a good idea to mic the amp as well if enough inputs and tracks are available. Not only does this provide more sonic options at mixdown, but you're also covered should one of the sources cut out in the middle of a song. Because retakes are impractical in most live-performance situations, ensuring that a song isn't ruined by an intermittent DI or mic cable is especially advisable.

#### **PRISTINE PATH**

Because any signal path is only as good as its weakest link, you must factor in the quality of the house equipment when deciding on a venue for a live recording. If you're getting feeds from the house console, you are dependent on the quality of that mixer, its mic preamps, and any other gear in the signal path.

Using a splitter gives you more control

gaffer's tape

tongue depressor

FIG. 6: Using a tongue depressor and nonresidue tape, you can attach a clip-on condenser mic to an acoustic guitar, allowing the performer to move around while maintaining a consistent signal.

over the quality of the signal path—in particular, the mic preamps. Some of the engineers I spoke with invested a great deal of money in mic preamps and felt strongly that the preamps were a critical factor in the sound of the final product.

High-quality cables also help maintain the best possible signal, as does keeping cable runs as short as possible. If you're going to get signals from the FOH console, try to position your gear close to the mixer, minimizing the length of cable runs.

#### **LEVEL HEAD**

Correct level setting is critical to the success of a live recording. Unlike recording in a studio, where you can do a second take should clipping occur, live recording requires that you get it right the first time. Therefore, extra care is required.

Sound check is your one chance to experiment with levels and settings, so it's helpful to have the band play as many songs as possible at that time. Since levels can change drastically from song to song, request that the group play, at minimum, both a quiet song and a full-volume number. Better yet, have them go back and forth between the two.

No matter how perfectly you set levels at sound check, the band will al-

most certainly end up playing louder during the show. Give yourself a margin of safety: after setting sufficiently hot levels, lower them a couple of decibels in advance. This is especially important when you're recording digitally, because signals above the maximum input level will necessarily distort. On the other hand, if the signal level is set too low, you lose fidelity (digital resolution) by virtue of not using the maximum number of bits; 20-bit and especially 24-bit recorders offer a big advantage here, thanks to greatly increased headroom.

Specific problems to anticipate include volume discrepancies (these can be huge) from one keyboard patch or guitar effect to the next. Hopefully, the musicians will have normalized volumes be-



Joel Singer, who has engineered many live recordings, emphasizes the importance of preparation to a project's success.

tween their different patches, effects, stompboxes, and so on, but you never know. Also, the human voice has a very broad dynamic range, so vocals can similarly be problematic. Therefore, talk to the band members before the show and question them in detail about their their mic techniques, their gear and how they use it, and so forth, to determine whether you might be faced with this type of problem.

Although keeping signal paths simple is desirable, often you'll need to patch in a compressor/limiter (or several of them, depending on the setup) to handle the big dynamic jumps common to live performances. Keep this to a minimum, though, because once you commit to tape with compression, there's no undoing it. In general, moderate settings will sound the most transparent. Try starting with a 2:1 ratio and setting the threshold so it triggers only on the loudest sounds. If the compressor kicks in too frequently or compresses the signal by more than 3 or 4 dB, lower the input gain.

Even with carefully set levels and a compressor guarding your signals, it's still imperative to keep a close eye on the input meters throughout the show. This is no time to space out or disappear for a smoke break. Recording live is very demanding in this regard—all it takes is one or two overshoots to tender a track unusable.





### PREPARATION IS KEY

Although the engineers I interviewed for this story often espoused different techniques, all agreed that diligent preparation is the key to live-recording success. The following guidelines are fairly universal and, if heeded, can do much to minimize gremlins.

Scout it out. The first step is finding a suitable location. Factor in not only the sound of the room but also whether the band is playing alone or sharing the bill. Unless you're contemplating a very simple setup, it's generally better to forego multipleband shows, due to both time and space limitations.

Once the venue is chosen, inspect it in advance. Visit the club to get a feel for how best to proceed; check out the power situation, the lay of the room, and, if it's a house sound system, the gear. Most important, find out who the sound person is and schedule a time for the two of you to talk. Do this in advance—if you show up a couple of hours before the gig with no notice, you're not likely to get a warm reception, and you may get no cooperation at all.

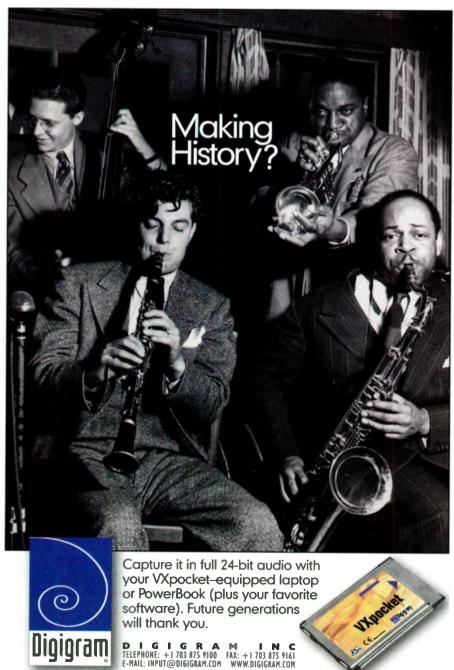
When talking to the sound person, explain clearly what you're doing and then discuss the equipment you'll be bringing and how it will interface (if at all) with the sound-reinforcement gear. As the sound engineer at Fez, a prominent New York City nightclub, Fred Reed has had a lot of experience dealing with bands coming in to do live recording. "Definitely get in touch with the sound person a week or two before the show," suggests Reed, "especially if you're doing an elaborate multitrack recording. If you're going to be coming out of the inserts, you'll need to make sure you're compatible, so find out what board the house is using, its pin configurations, and so on."

Philip Perkins concurs. "There's no substitute for going to the venue and talking to the people who work there," he says, "especially if you're going to interface with the house sound system. Pay the engineers a visit or at least talk to them on the phone. Most people, if they feel like you're including them, will want to help you do a good job."

Keep it clean. Keeping your recording equipment well maintained is important. Inspect every piece of gear carefully before leaving for the gig, including mics, headphones, cables, and connectors. If possible, plug everything in to make sure each item in the signal chain is fully functional.

You should always bring along spare parts as well, including cables, connectors, power strips, power cords, extension cords, mic stands, mic clips, stereo bars, recording media, duct tape, and any other small, affordable essentials. Because you may be working with a questionable house system, having your act together is a necessity—doubling up on the small stuff can really save the day.

Test it out. Not only is sound check important for setting levels, but it's also the sole opportunity you'll have for a "dry run" of your setup. Record the entire sound check and then listen



back afterward for problems. Examine each track individually to make sure you're getting a good sound and to check for distortion, hums, and buzzes. This is especially important if you aren't set up in an isolated location. Unless the band is playing more than one set, this is your last opportunity to hear what you've recorded until after the show is over.

Fight the power. Some of the nastiest gremlins to be devil live recordists are the buzzes and hums caused by ground loops and other AC problems. Try to resolve such problems before the show, or else the recording could be jeopardized. Typically, it's best to power your gear from the same leg of the club's AC power that the sound system is on.

Use power conditioners to lessen the effects of noisy power lines, and allow some time (during or after sound check) for changing polarities, lifting grounds, and troubleshooting power problems.

Plug the leaks. Leakage is another problem you'll face as a live recordist. Whereas studios can use iso booths, baffles, gobos, and other sound absorbers (not to mention the most surefire weapon, overdubbing) to help keep sounds from leaking onto other tracks, the live recordist typically lacks those options. Directional mics and Dls can minimize leakage, but some bleed will inevitably occur.

Stage monitors are often a cause of leakage. Vocal and overhead mics are particularly prone to picking up sound from the monitors, which can lead to real problems during mixdown (for example, the vocals leaking into the drum overheads). Another problem is that singers (and sometimes other performers) typically request more and more level from the monitors as a show progresses. Therefore, it's important to explain the situation in advance and, ideally, persuade them to use as low a monitor level as possible. Making a successful recording is in their best interest, so they're likely to be receptive to your suggestions.

### **PARTING WORDS**

After the show is over, the gear has been packed up and brought home, and you've gotten some rest, plenty of work remains to be done. A 2-track recording won't require mixing, of course, but it will almost certainly benefit from some editing and mastering.

With a multitrack project, however, you have the option of rerecording certain tracks, overdubbing extra instruments, and the like prior to mixdown. In other words, your work has just begun.

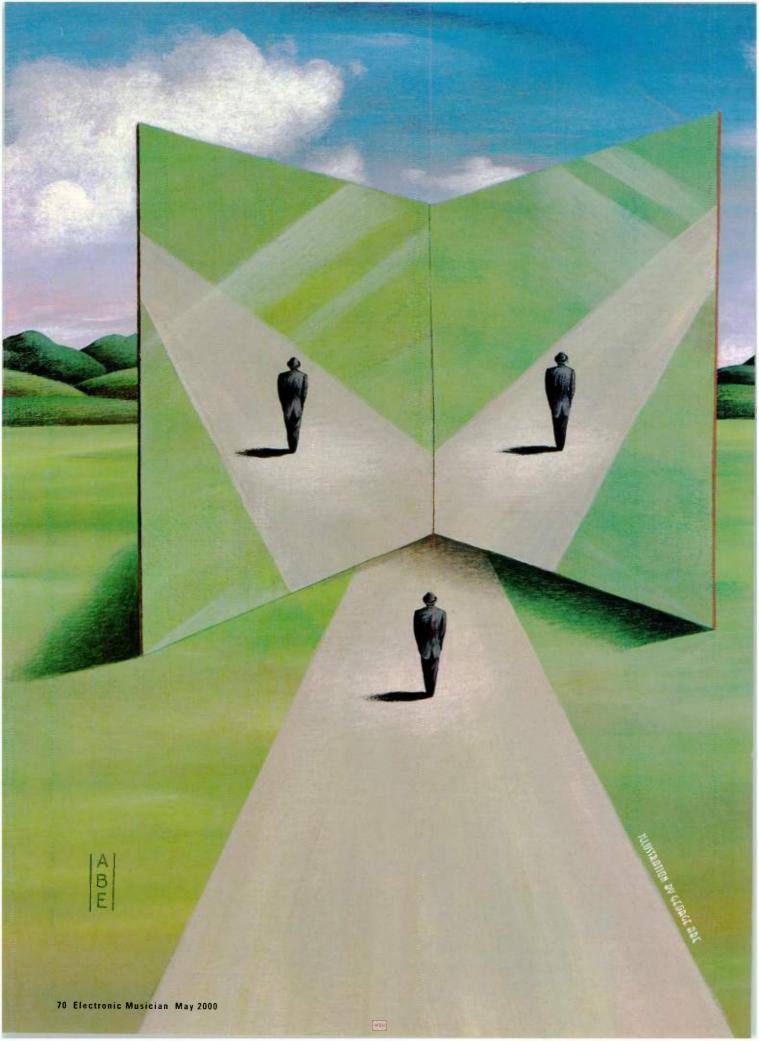
An automated mixing system such as Pro Tools can be extremely helpful for fixing some of the problems inherent in live recordings as well as for enhancing the overall sound. You can adjust fluctuating levels, mute open mics in parts where they weren't used, and generally make the record-

ing as clean as possible. Equalization and effects can also be applied as needed.

As long as the changes you make serve the project, don't hesitate to experiment. After all, there are no absolute rules of recording other than, "If it sounds good, do it."

Mike Levine is an associate editor for EM and editor of Onstage, EM's new live-performance quarterly magazine. When he's not buried in his word processor, he also composes music for commercials.







# THE SPLITTIN'



CREATE GREAT

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FROM MONO SAMPLES.

Even if your sampler has 128 MB of RAM or more, sometimes you can't or don't want to use stereo samples. Stereo samples use at least two voices of polyphony per note—more if you're using multiple crossfaded layers.

A stereo instrument with three velocity-crossfaded layers consumes 6 voices per note; a five-note chord eats 30 voices. If the instrument has a long release time or you repeatedly use the sustain pedal, you can easily have 90 or more notes going at one time. Even if your sampler has 128 voices of polyphony, you're left with considerably less room for other instruments, especially if they, too, are in stereo.

In addition, stereo samples use twice as much sample RAM as mono samples of the same instrument. This isn't a big problem in the studio because you load an instrument into RAM only when you're ready to lay down its track. But in a live performance, you may need to have hundreds of different instruments ready to go on demand.

By Daniel Fisher

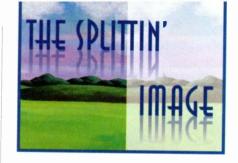
# IMH(E

What are your options if you like a dramatic stereo effect but can't afford the resources that you need to use stereo samples for every track? And what if your samples are available only in mono, like those in ROM-based synths and samplers?

#### **EXPANDING YOUR WORLD**

Fortunately, there are several ways to create a satisfactory artificial stereo image from a mono sample. But will it really be satisfactory? Yes, sometimes. Your ears and brain can be very picky about the precise nature of a stereo image. For instance, a listener is more likely to detect an artificial stereo image of a single instrument. But when it comes to simulating a pleasant stereo ambience, you can often just "throw things together," and the brain will quite happily interpret the results as a nice, wide stereo field.

Each method of generating a stereo sample from a mono source has good and bad points. I'll discuss some of the more common methods, then share a very successful technique that I developed for some of my CD-ROM projects.



Although this article focuses primarily on techniques to be used with samplers and ROM sample-playback synthesizers, many of the tips are equally useful for any type of monaural audio source. For example, you can try some of the tricks on a monaural live recording or mono audio track from a video.

#### **BACKWARD COMPATIBILITY**

Let's consider some possible consequences of using artificial stereo images. Our pseudostereo sounds will be wonderful when played on stereo systems, but what if, after all that work, they are played back in (shudder) mono? They still need to sound good, of course, and achieving that is not always simple.

Why would they be played in mono? AM radio stations and many house P.A. systems in clubs and arenas are mono. In addition, certain effects processors—both external units and onboard effects in synthesizers and samplers—sum the left and right inputs to mono before they apply a stereo effect. The odds are good that your beautifully crafted stereo mix (or stereo sample) will get played in mono somewhere along the line. So when you're creating a stereo image, it pays to consider mono compatibility.

This is easier said than done. Unwanted sonic artifacts often appear when an artificial stereo image is collapsed down to a monaural channel. One common artifact is comb filtering, which gives the sound an unnatural, tubelike character or creates annoying peaks or notches in specific frequency bands. Another possible problem is the loss of clarity, or even a complete cancellation of the signal. So when you create your pseudostereo image, you must be very sly.

#### THE USUAL SUSPECTS

Here's a quick roundup of seven common image-manipulation techniques for making a stereo sample from a mono source. I'll note the primary benefits and drawbacks of each method.

Panning frequency bands. This is one

of the more delicate (that is, less intrusive) ways to expand an image. Take a mono instrument sample and separate it into four or more frequency bands. Pan each band, alternating between the left and right sides. Try all panning values for each frequency band, and experiment with the center frequencies and bandwidths if those parameters are available. Make sure you occasionally pan your mixer to mono to hear what the left and right channels will sound like when blended.

This effect works best if you can use four or more bands. (With only three bands, the highs and lows would be panned left, leaving only the mids to go to the right.) If you do this properly, the stereo illusion's quality will be determined mostly by the bandpass filter's phase accuracy and control over its bandwidth.

Delaying or detuning one side. Two of the most common methods of creating an artificial stereo image on synthesizers and samplers are delaying a copy of a sample relative to the original version and detuning a copy relative to the original. You can also combine the two techniques.

There are several ways to use delay. One method is to assign separate but identical instrument layers to the left and right channels. Delay playback of one side via the sampler's Start Delay parameter, then pan the channels to opposite sides.

Another method is to split the mono signal into two identical signals, delay one signal relative to the other, and pan the dry (unprocessed) and wet (delayed) signals to opposite sides. One common way to do this is to send the mono signal through a digital delay line (DDL) set to 100 percent wet (processed). If the DDL has separate dry and wet outputs, you can split the mono signal that way. Otherwise, you'll have to split the signal before the delay by, for instance, sending the same mono sample to the left and right synth outputs.

You can also use a sample-editing program. Transfer the sample to your computer and open the mono original on the left track of a stereo file. Copy the sample to the right track. Trim or insert silence at the beginning of the copy, phase-shifting it relative to the original (see Fig. 1). Save the file for future use, and fly the new stereo sound back into your sampler.

All of these approaches accomplish the same thing: they delay playback of a copy of the sample. You just have to decide whether you prefer to sacrifice additional polyphony to the second layer in your sampler, use a signal processor (the delay), or muck about with editing on a computer.

Although popular, the delay method is not without flaws. If you're using a start delay (or delay time) between 2 and 20 milliseconds, you could end up with comb-filtering artifacts when the signals are combined to mono. Delay times longer than 24 milliseconds tend to result in fewer comb-filtering artifacts, but the signal will increasingly sound like a rapid slapback echo as you lengthen the delay time.

In addition to adjusting the time difference between the left and right sides, you can slightly detune one side

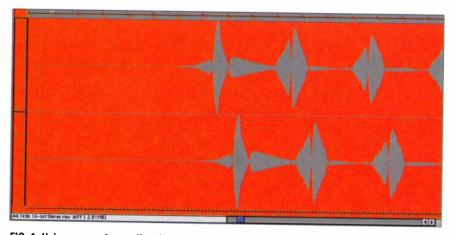


FIG. 1: Using a waveform editor (BIAS *Peak* 2.1 for the Mac), I created a 2-channel sound file with identical copies of a mono bass-guitar sample on each channel. I then delayed the right channel by seven milliseconds relative to the left channel. I could have taken the technique further by slightly detuning the two sides in opposite directions.



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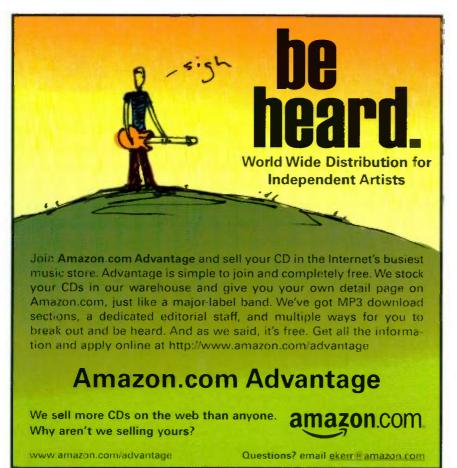


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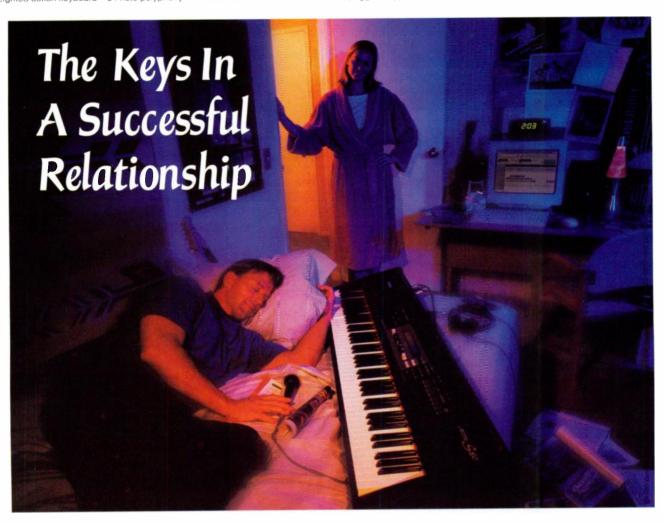
relative to the other. Usually you do this in the Pitch or Oscillator page of your synth or sampler. You can subtly detune by raising or lowering the pitch of one side by one or two cents. If you want a more noticeable offset, raise the pitch on one side and decrease the pitch on the other side by the same amount. This will help keep the overall pitch centered instead of sharp or flat.

For an even more animated quality, assign an LFO to the pitch of one or both sides. Again, by making the pitch respond positively to the LFO on one side and negatively on the other, you'll keep the average pitch centered.

Phase inversion. Of all the techniques used to create a stereo image, this produces the most unnatural artifacts. As with the "delay one side" trick, you can do phase inversion with two instrument layers or with a digital delay that allows you to invert the phase of the processed signal while keeping the delay time at 0. Once again, you'll pan the dry and wet signals to opposite sides. Two considerations here are whether you can afford to use twice as much polyphony and whether you have a processing block that can invert the signal phase.

The end result is a dramatic, hyperwide stereo image that, depending on your point of view, is either radically unnerving or sonically irritating. When you listen through headphones or sit right in the sweet spot, you'll feel as if the insides of your ears are being pulled away from your head. And you can forget about summing to mono. If the two sides are truly 180 degrees out of phase, the mono version will be silent because the two sides will cancel each other out.

Different sample-start points. If you have a long, ambient, textural sample, or a sample of a complex analog synth oscillator, try this trick. Create a copy of your ambient sample (such as the sound of rain, a bubbling stream, or a vinyl record spinning around) and make a new start point one-third to one-half of the way between the original start point and the loop point. Now,



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when you press a key, one side of the stereo signal will start at the beginning of the ambience and the other will start somewhere in the middle. Experiment with panning the two sides, and you'll come up with a lively and highly dimensional sound that remains

interesting even after repeated loops.

By using this technique on a sample of a thick analog oscillator, you can create a huge sound with subtle and intriguing stereo motion. Try it on a sample of white noise—you'll perceive a very complex stereo image that doesn't seem to repeat.

If your synth or sampler can choose sample-start points at random, you can get even more interesting results. Again, experiment with wide and narrow panning. (If the panning is too wide, the sound will seem unnatural.)

When you find the right spot, you'll hear elements that seem to exist at points all over the stereo field.

Different sample-loop points. This is similar to using different sample-start points, but the stereo perspective changes while the samples are looping. Simply make a copy of the original sample and find somewhat shorter loop points, so that when the shorter loop is hard-panned against the original sample, you have a complex stereo interaction that plays for a long time before the cycle repeats. The use of different start and loop points usually translates well even if the two sides are panned all the way back to mono.

Different start and loop points are effective on a sample of white noise because, by definition, the spectral content of white noise is random. The fact that a 2-channel sample of white noise repeats itself can be disguised by the interaction between the left and right signals. You can then filter this stereo version to create the ambient sound of thunder, wind, waves, rain, and other noise-based sound effects.

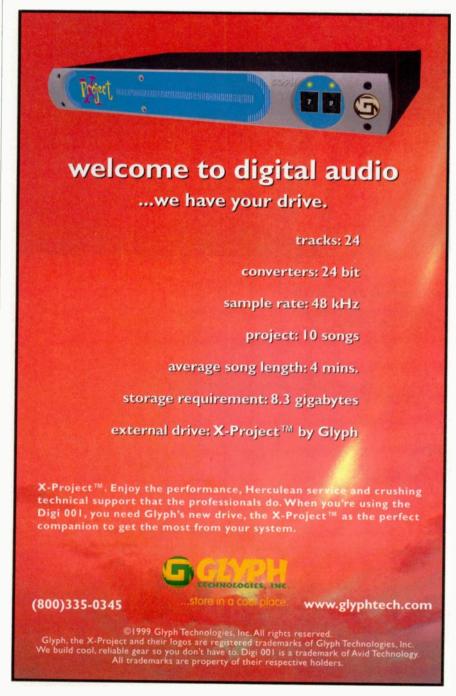
Different instrument samples. This technique is most successful with ensemble samples. Let's say you have two different sax-ensemble synth instruments. By using two layers and carefully experimenting with each layer's panning, volume, and tuning, you can achieve a believable and lush stereo ensemble that is far more pleasing than either ensemble alone.

The key is to keep trying different values for each of the parameters until you find the sweet spot. The values may be different for each keymapped sample range. Use your ears to create a uniform ambience all the way up and down the keyboard. A stereo image generated in this way can almost always be collapsed to mono with minimal artifacts.

Short reverbs and room effects. You can create a full-sounding stereo image by using either a "short" room setting on a room simulator or just the early reflections parameter on a digital reverb. But you're likely to hear artifacts if the signal is combined back to mono.

#### **MANUAL SHIFTING**

Manual timbre-shifting is not a stereoimaging trick. It is a method of altering the playback of a monaural, multisampled instrument in such a way that the edited instrument exhibits timbral





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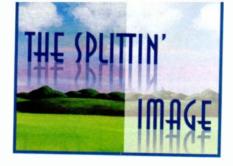
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characteristics that are shifted in both pitch and time; the end result is not an artificial stereo instrument but rather an enhanced monaural instrument. The process involves remapping the multisamples up or down to the next keymap range and then modifying their playback rate so that the samples play the correct pitch. Some samplers and sample-playback synths have a dedicated parameter for doing this, whereas others accomplish it through several related parameters. Unfortunately, some synths won't let you pull off this trick at all.

Here's how it works. Let's say you have a piano multisample with a different sample for every whole step. You might have a sample at C4 and another at D4, E4, F#4, G#4, A#4, C5, and so on, up and down the keyboard.

Manually timbre-shift this instrument by transposing it up a whole step and then lowering the pitch by 200 cents (or a whole step) in your synth's Pitch page (see Fig. 2). Now, pressing the C4 key

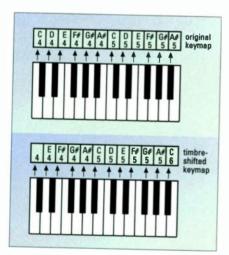


FIG. 2: I've enhanced a sampled instrument by manually timbre-shifting. Pressing the C4 key on a MIDI keyboard now triggers the D4 piano sample instead of the original C4 sample. The pitch needs to be lowered by 200 cents so that you still hear the correct C note. However, this process only alters the shifted sample's frequency, not its formant and time base, so it will sound slower and darker than the original C4 sample. Manually timbre-shifting in the opposite direction yields a brighter sound.

actually plays the D4 piano sample, but because you decreased the pitch by 200 cents, you'll still hear the correct C note. However, you have only changed the lowered sample's frequency, not its formant or time base, so it will sound slower, deeper, and heavier than normal. Conversely, if you transposed the piano down a whole step and then raised the pitch by 200 cents, the result would be a piano with a slightly faster, brighter, and thinner sound.

To create an artificial stereo ambience with this timbre-shifting technique, simply pan an unaltered version (that is, playing C4 triggers a real, unshifted C4 sample) to one side and a timbre-shifted version to the other side. By experimenting with upward and downward timbre-shifting, the number of half steps that the instrument is shifted, and the depth of panning, you can get a wide variety of stereo images that also contain a much fuller sound.

Timbre-shifting does have drawbacks, though. For one thing, you need to tune each sample's pitch very carefully or the two sides will chorus. Furthermore, the instrument's pitch must be stable over time, because differences between the left and right layers will be very apparent. Finally, many instruments do not have the same number of notes in every key range. For example, an instrument might be sampled every six notes in the bottom range, every two notes in the midrange, and every octave in the extreme high range. Thus, timbre-shifted notes may act differently in the various ranges, leading to an uneven stereo image across the keyboard.

#### HANDCRAFTED BEAUTY

Handcrafted stereo timbre-shifting is a method that I developed over the past five years while programming roughly 24 sample CD-ROMs. I put the technique to full use on a guitar CD-ROM that includes 115 different acoustic and electric guitars and basses. All of the guitar samples that were originally recorded in mono also feature stereo versions that you can sum to mono or pan to any desired position in between without generating artifacts.

The process is based on the timbreshifting method described earlier, but instead of simply shifting a mono sample, you create two identical layers (as with some of the other techniques I've

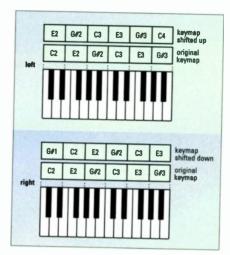


FIG. 3: Start with a mono string ensemble. Create a 2-channel version that is identical on both sides. Then shift each keymap on the left channel to the next highest sample (a major third above the original); for instance, playing C2 on the keyboard now triggers the E2 sample. Adjust each left-channel sample's tuning down a third. Fine-adjust each sample's volume and panning to complete the job. For best results, timbre-shift both copies in opposite directions, as shown here.

discussed) and shift the timbre for one or both channels. After that, you can individually fine-tune the samples in every key range, or even every key. The resulting artificial stereo field is pleasant and consistent over the whole keyboard. The technique works equally well on ROM- and RAM-based sampled instruments, but for the best results, your instrument multisample should have more than two samples per octave so that the samples won't get shifted too far from their normal pitch range.

#### **SHIFTY STRINGS**

Let's start with a string ensemble that has been sampled three times per octave, with the sample roots at every C, E, and G# throughout the instruments' range. For now, we'll assume that the root keys are at the bottom of each key range, so every C sample is pitch-shifted to cover the range from C to D#, every E sample covers E to G, and every G# sample handles G# to B.

Make a copy of the keymap; name one copy String Ens L and the other String Ens R. String Ens L will be in our program's first layer, panned hard left. String Ens R will be in the second layer, panned hard right.

Next, edit String Ens L, starting with the first sample key range. Let's say that

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# Bit by Bit

#### The truth about digital audio transfers.

By Brian Smithers

s more and more sound cards support digital audio transfers, it's important to understand what various audio interfaces may (or may not) do to your carefully crafted sounds. The primary objective of a digital transfer is to create a bit-accurate clone of a digital recording (as when you copy a DAT tape to your computer or vice versa). However, a number of devices with digital interfaces can only perform this task under certain circumstances, and some can't do it at all.

If you find this situation a bit confusing, you're not alone. I was surprised

by the amount of misinformation I came across while researching this column. But don't fret—it's not that complicated if you're familiar with the different digital audio formats and their characteristics.

Last month, we examined the four most common types of digital audio formats and described the features and capabilities that each one provides (see "Square One: Spare Interchange" in the April 2000 issue of EM). This month we'll look at some real-life situations and describe how you can avoid common problems when making digital transfers.



Let's briefly recap the primary formats for exchanging digital audio data. Digital interfaces currently come in two types: stereo and multichannel. Each has two major formats. The 2-channel formats, AES/EBU and S/PDIF, use different cables and connectors, although both use the same method to encode data. In addition to using traditional coaxial audio cables, S/PDIF can also transmit over fiber-optic cable. The two multichannel formats, ADAT Optical (Lightpipe) and TDIF, can each transfer eight channels at once.

At the dawn of the digital age, the bit-accurate piracy of copyrighted works loomed large, leading to the requirement of a form of copy protection in digital interfaces that target



iam. You've played for d play a junior prom if top ten radio from the You got a band. Or maybe just a guitar and your mom. You've played in dive bars. He they let you rock. You know every lyric plast fifteen years but you can't remember the hair, the tattoo, the attitude, the tate you're destined for greatness... destined state capitols. You've got ind the desire. You know be huge. You just need a little exposure.

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consumers. The Serial Copy Management System (SCMS, fittingly pronounced like scums) defined two hidden flags in the S/PDIF data stream that prevented consumer machines from making digital copies of digital copies. AES/EBU, the professional interface, carried no such burden. Today, consumer DAT decks are essentially extinct, and SCMS is all but forgotten, so most S/PDIF devices ignore it completely.

Although issues surrounding SCMS have faded into the background, other problems remain when different types of digital audio devices attempt to communicate. Therefore, the first rule of digital transfer is knowing which format your gear supports. Check your connectors: each format uses a different type of connector, with some exceptions—ADAT Optical and S/PDIF optical use identical fiber-optic connectors—so you need to know which optical format you're dealing with.

The proliferation of format converters makes it easy to connect different formats. For example, Tascam makes a device called the IF-TAD that converts from TDIF to ADAT Optical, and you can also use MOTU's 2408 or Soundscape's SS8IO-2 as stand-alone TDIF-to-Lightpipe converters. For stereo transfers, Midiman's CO3 transmits your signal between any combination of AES/EBU, coaxial S/PDIF, and optical S/PDIF devices.

#### **GUMMING UP THE WORKS**

It all sounds so orderly—if A plugs into B, you're set to clone, right? That's what I expected when I tried to use my Opcode Sonicport (S/PDIF-to-USB connection device) to transfer some tracks from my notebook to my desktop computer. I connected the coaxial S/PDIF output of the Sonicport to the coaxial S/PDIF input of my desktop sound card, punched Play on one and Record on the other, and watched helplessly as everything arrived 6 dB quieter than it started out.

We've all chuckled at the phrase "It's not a bug, it's a feature!"—but this time it's true. The decibel attenuation of my tracks makes sense when you consider that from a consumer perspective, USB audio is intended more for connecting speakers than for recording. Therefore, the USB audio fader goes into the Windows Mixer applet with a default setting of 6 dB

below maximum to avoid blowing out USB speakers. All I had to do was raise that fader to its maximum setting to make my Sonicport send in the clones.

Mixers, outboard A/D/A converters, effects, and other devices with digital interfaces are handy when you want to have control over input and output gain. If your goal is to produce bit-accurate copies, you need to override the gain controls or set them to the proper level. Often this simply means setting a software fader to unity (0 dB). Check your device's manual

or consult the manufacturer for the appropriate setting.

Guillemot has simplified this process for its Maxi Studio ISIS card. The mixer applet features a simple check box labeled Backup. The Backup option disables the S/PDIF output faders to ensure that your mix doesn't suffer any unintended gain adjustments. Here's proof that intelligent design doesn't need to cost an arm and a leg.

#### **WHO'S IN CHARGE HERE?**

It's a fact of digital life that no two clocks agree completely. This can cause glitches when you're transferring digital audio. If the receiving device's clock runs faster than that of the sending device, it will eventually expect to receive a sample that you haven't sent yet. If it runs slower, it will need to drop a sample to keep up. These discontinuities in the data stream appear, respectively, as flat spots or jumps in the waveform and often cause audible clicks (see Fig. 1).

You can avoid this problem in one of two ways. The most common solution is to designate one device as the master clock, so that the slave device receives timing information from the master. This leaves no room for disagreement between the two devices and ensures accurate transfer of the waveform. Most sound cards refer to this master-slave relationship in terms of internal-external clock or sync. Set the master device to use its internal clock and the slave device to receive timing information from an external source. If your interface doesn't offer an external option, it probably can't be a slave to

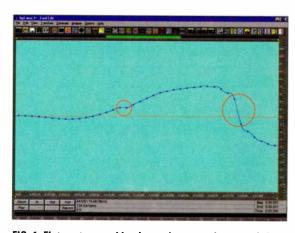


FIG. 1: Flat spots or sudden jumps in a waveform result from missed samples. Avoid this by setting the proper master-slave relationship between the two devices or by synching both to an external clock source.

another device and therefore won't do bit-accurate transfers.

The high-end solution to synchronization is to slave all devices to a superaccurate external clock. If your gear has word-clock or Superclock inputs, that's what they're for. Of course, this requires that everything else in your studio be capable of operating in slave mode.

E-mu has an interesting approach to this master-slave relationship. Its Audio Production Studio (APS) operates only on an internal clock, which is fixed at 48 kHz. Any audio coming through the APS device's S/PDIF port undergoes a sample-rate conversion to 48 kHz and another sample-rate conversion if you're saving it as anything other than a 48 kHz file. (To avoid piling up conversions, E-mu advises users to work exclusively at 48 kHz and then convert the final mix to 44.1 kHz before burning a project to CD.) Because this conversion affects the sound less than an analog conversion would, the APS manages to make high-quality digital copies but not bit-for-bit clones.

#### A BIT OF CAUTION

Naturally, it's important to make sure that your source and destination devices are set to the same word length—and that they can handle it. Try sending a 24-bit signal to a device expecting only 16 bits, and you could get an ugly surprise. Also, never assume that all digital interfaces support the same resolutions. All four formats can pass 24 bits, but not all manufacturers implement the formats that way, even in their newer, high-profile products.



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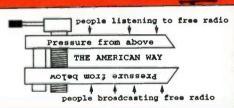


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#### DESKTOP MUSICIAN

The ISIS, for example, only supports 16-bit transfers through its S/PDIF interface. For many purposes, this is not a limitation, but if you're hoping to transfer higher-resolution audio, you need to buy a card that supports the format's full bandwidth.

Since current ADATs are 20-bit devices, some ADAT Optical interfaces don't support 24-bit transfers. One particularly high-profile device that passes only 20 bits is Digidesign's ADAT Bridge I/O for TDM systems.

If you have an older S/PDIF-equipped DAT deck with SCMS, you'll want a digital interface that knows how to deal with the extra bits. TerraTec's audio cards, such as the EWS88-MT, give you the option of keeping or ignoring the copy protection bits. Of course, this isn't an issue with an AES/EBU interface or with either multichannel format.

#### **CHAMPING AT THE BIT**

To paraphrase Murphy's Law, things can and will go wrong. For example, if your DAT or MDM encounters errors on the tape, it will try to correct them. Other than maintaining your decks and taking proper care of your tapes, you can't do much about this.

Once the data has left the source device, bad cables or interference can

#### **CLONING 101: ENSURING BIT-ACCURATE TRANSFERS**

To transfer bit-for-bit clones consistently between digital devices, keep the following checklist handy:

- 1. Check your connections; make sure that they're snug and secure.
- 2. Use the shortest possible run of the highest-quality cable.
- 3. Make sure that you have both devices set to the same bit depth and sample rate.
- Set the source device to its internal clock (master) and the destination device to external clock (slave).
- 5. Disable input and output faders if possible; if not, set them to unity.
- 6. Test your results in an audio editor to confirm that accurate transfers are occurring.

corrupt it in transit. Installing quality cables, maintaining them properly, and using the shortest cable runs possible limit opportunities for this sort of error (see the sidebar "Cloning 101: Ensuring Bit-Accurate Transfers" for some additional tips).

You can easily test the reliability of your digital transfers by comparing files in an editing program. Transfer a file from your computer to DAT (or to another computer with a matching digital interface), then transfer it back. Some audio editors offer a function that compares two files and reports any differences. You can do this manually by inverting one file and pasting it over the other (see Fig. 2). Phase cancellation causes two identical files to leave a straight line on 0 dB. Anything left over

is an error introduced in the transfer process. If you see a miniature version of the original file or its inverse, you have a level mismatch. Periodic bumps in an otherwise straight line could indicate a clock problem, such as an improperly set master-slave relationship. Nonperiodic remnants may result from bad connections or line interference.

If you're shopping for a digital transfer device, consider a few basic questions. How important is bit-accurate transfer to you? Does the device support the sample rates and bit depths that you require? Is it capable of slaving to an external clock? Some of these questions may require researching the manufacturer's Web site and possibly even e-mailing or calling tech support with specific queries.

#### **FINAL TIDBITS**

Digital transfer may not be the best way to move your audio around, at least between computers. You could burn an audio CD, then use a ripper program to retrieve the track, or you could save the file as a data file to a CD-R or a Zip or Jaz disk. Ethernet connections are becoming more popular in personal studios because they facilitate the sharing of large audio files. I've even used my notebook's infrared port to share audio files with others.

Nonetheless, if you have digital tape in the mix, you'll want reliable digital transfers. Evaluate your needs, run some tests, compare the results, and adjust accordingly. Once you've worked it all out, write down the procedure so that you won't forget a step when the heat is on. When cloning time comes, you shouldn't need a bit of luck.

Brian Smithers is studying the effects of 24-bit recording on 2-bit performances. Share your thoughts with him through his Web site at http://members.aol.com/notebooks1.

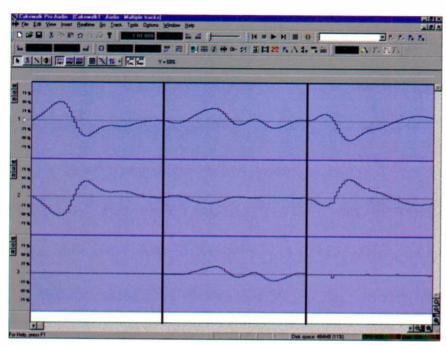


FIG. 2: By inverting a file and pasting it over its clone, you can determine how accurate the transfer was. In this illustration, the first column shows the result with a perfect clone. The second column shows a good copy at a different volume. The third column shows a periodic glitch, probably the result of poor synchronization.

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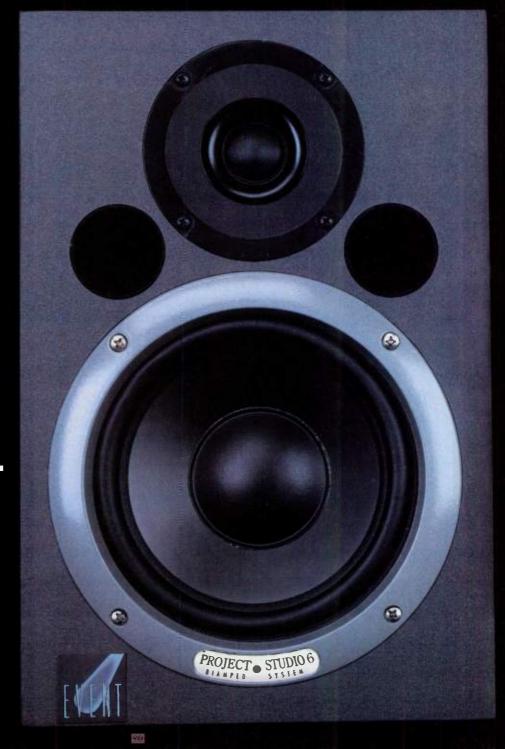
And then Frank and Walter created the Project Studio 6—and gave it low end response that simply blows the laws of physics to smithereens. A 6.5" speaker in a tiny cabinet producing a deep rich 42Hz... astounding!

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## Downloadable Sounds

An endless supply of sounds is just a click away.

By Jennifer Hruska

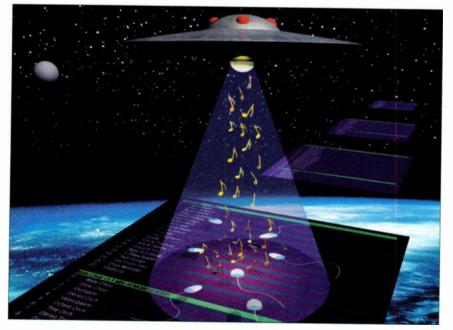
one of us want our hardware to become obsolete. So what do you do when you've tired of your sound card's ROM presets or you can't stomach another General MIDI violin? You might find the answer in downloadable sounds. These give you access to hundreds or even thousands of new sounds that you can use in your sound card, in a hardware sampler, or even in a software sampler running on your computer. Unlike with most sound-card presets, you can tweak, twist, and fine-tune downloadable sounds to your liking.

Here, we'll discuss SoundFont and DLS, the two most common formats for downloadable sound, as well as thirdparty editors and libraries for creating your own sounds. We'll also cover some of the hardware and software supporting these formats and explore their use in your music productions.

#### THE SOUNDFONT SPECS

SoundFont is a downloadable-sound format designed by Creative Labs. Originally limited in their specifications and use, today SoundFont files can sound great, and a number of Windows and Mac audio programs support them. SoundFonts consist of Presets (similar to Roland Patches or Kurzweil Programs) that are assigned MIDI Program Change numbers with which you can select them. Presets are made from Instruments, which contain voicing data (more on this later) and keymap information that links samples to specific notes on the keyboard. The keymap consists of samples drawn from a collection called the Sample Pool.

SoundFonts typically have an SF2 file extension. The format combines two types of data: the first is the sampled audio data itself, typically originating as a WAV file, and the second is the synthesis information required to articulate or modulate the digital audio, & sometimes referred to as the voicing \( \bar{2} \) data. When you strike a note on the keyboard, a MIDI Note On message h





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goes to the SoundFont synthesizer, which tells it the corresponding sound to play (sampled audio data) and how to respond over time (voicing data).

The sampled-audio specification calls for standard 16-bit samples with a variable sampling frequency of up to 48 kHz. Samples use a standard looping scheme comprising parameters for Sample Start, Sustain Loop End, and Sample End. Other parameters include Original Key, which defines the MIDI key number that the original sample corresponds to, and Pitch Correction, which allows a tuning adjustment of ±100 cents.

The voicing data falls into two distinct categories as well, function blocks and function modulators. Function blocks are the characteristics of the sound that you can control—frequency, pitch, and amplitude. Function modulators are the parameters that affect these three characteristics. Modulators can consist of internal sources such as low frequency oscillators (LFOs) or envelope generators (EGs); physical sources such as the modulation or pitch-bend wheels; or MIDI control sources such as

MIDI Controllers 91 and 93 (Reverb Depth and Chorus Depth). The modular design provides a great deal of flexibility—you can route any modulation source to any modulation function, resulting in greater sonic variety.

SoundFont internal modulators consist of two EGs, one tied to amplitude and one available for pitch or filter modulation. Each EG allows for standard Attack Rate, Decay Rate, Sustain Level, and Release Rate, plus a Delay Rate segment prior to the Attack Rate and a Hold Rate segment between the Attack and Decay segments. Two LFOs are available to modulate frequency, pitch, or amplitude. Each LFO consists of a Frequency Rate that determines the speed of the oscillator and a Delay Rate that controls the time until the LFO starts its modulation.

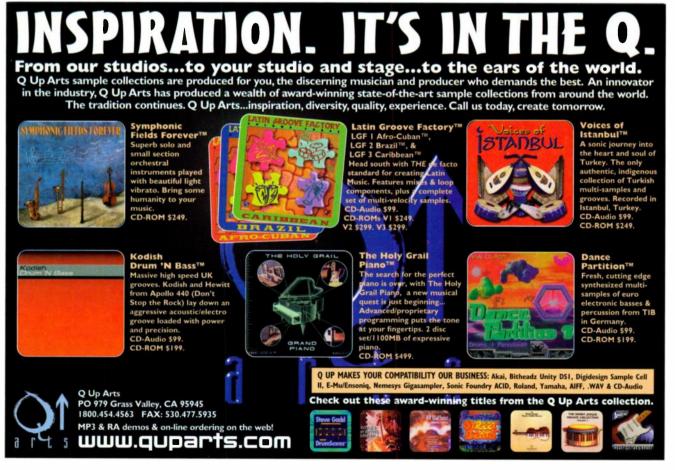
A SoundFont can also include a filter, typically a second-order (2-pole) filter with resonance. Its frequency range is 0 Hz to 20 kHz. As previously mentioned, SoundFont effects include Reverb Depth and Chorus Depth, which respond to MIDI Controllers 91

and 93, respectively. Because the synthesizer generates the effects themselves, the quality varies from device to device.

#### **DLS LEVEL 1 AND DLS LEVEL 2**

DLS stands for Downloadable Sounds. Like SoundFont, DLS is a sound-file format. The two current versions of the DLS standard are known as DLS Level 1 and DLS Level 2. The MIDI Manufacturers Association (MMA) developed both in collaboration with members of the Interactive Audio Special Interest Group. For detailed specifications on the DLS Level 1 format, see "Desktop Musician: Down and Out in Cyberspace" in the December 1998 issue of EM, or visit the MMA's Web site at www.midi.org. The following passages provide a brief overview of the DLS structure as well as a description of the additions featured in DLS Level 2.

DLS files consist of Instruments, Regions, Articulations, and the samples themselves. Instruments—equivalent to SoundFont Presets—have assigned names and MIDI Program Change numbers for easy access. Two types



of Instruments are available: Melodic Instruments, which contain up to 16 Regions, and Drum Instruments, which contain up to 128 Regions. A Region includes keymapping information and a reference to the audio sample to play. Regions also contain the synthesis, or voicing, parameters-DLS calls them Articulations-such as EGs and LFOs. DLS Level 1 Articulation parameters consist of two Envelope Generators, one for pitch and one for amplitude, as well as one LFO for modulating pitch or amplitude. Melodic Instruments can contain one Articulation for all Regions. whereas Drum Instruments can have a separate Articulation for each region. DLS samples can consist of 8- or 16-bit sample data and contain pitch, volume, and root-key information.

DLS Level 2 provides increased functionality to the DLS Level 1 specification. Additions include a 2-pole filter with resonance, a second LFO, layering of Regions, 6-stage Envelopes, per-Region Articulations on Melodic Instruments, and standardized responses to MIDI controllers. DLS Level 2 also includes a standardized method of adding extra synthesis capabilities for developers who want more features.

#### SOUND RESOURCES

You can choose among several different sound editors for making your own downloadable sounds. Creative Labs includes Vienna, a popular Sound-Font editor, free with its SoundFontcompatible cards (see Fig. 1); you can also download Vienna from the Web at www.creativelabs.com. Another popular editor is Awave (\$55) by FMJ-Software (www.fmjsoft.com). This generalpurpose sound editor supports multiple file formats, including SoundFont, DLS Level 1, and DLS Level 2. Audio Compositor (\$40), another shareware editor, supports SoundFont and both levels of DLS; you can find it at http:// home.att.net/~audiocompositor. Megota Software offers a SoundFont patch manager called AweVBank 98 as well as a SoundFont compression program called SFPack, both available at www.megota.com.

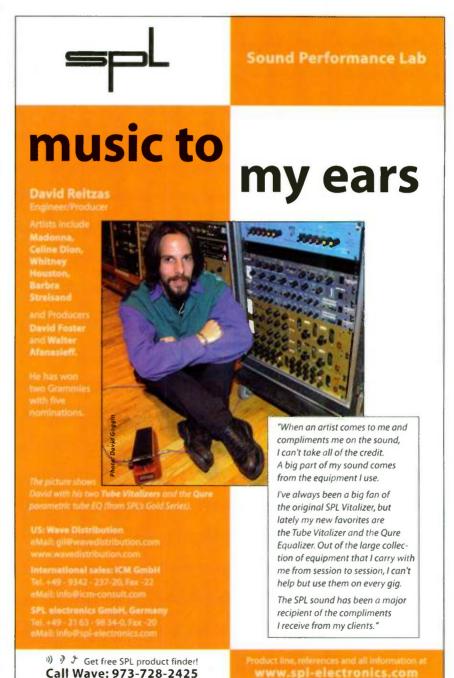
Steinberg's ReCycle 1.7 allows you to dissect audio files into samples and MIDI events, and it has editing features similar to those of Vienna. You can get more information at www.us.steinberg .net. Finally, Microsoft's Synth Author and

DirectMusic Producer are free DLS Level 1 editors available at www.microsoft.com. (DirectMusic 8.0, which Microsoft will release this spring, allows DLS Level 2 editing and auditioning.)

The number of hardware and software synthesizers that support Sound-Fonts and DLS is rapidly growing, no doubt due to the synthesis power that they can generate at such an attractive price. Some of the more popular sound card-based synthesizers supporting SoundFonts are the Creative Labs Live card, the E-mu Audio Production

Studio (APS), and the EWS64 line of TerraTec cards (very popular in Europe). These cards have much higher specs than the older models do and often include studio-quality  $\Lambda/D/\Lambda$  converters, balanced analog I/O, S/PDIF I/O, and high-quality stereo and surround-sound effects. For DLS-supported sound cards, check out the Turtle Beach Montego II, the Yamaha 192XG, and the TerraTec line.

Software-based synths and samplers are convenient because they interface with whatever audio output device you



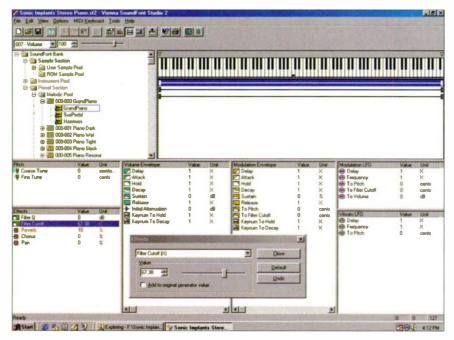


FIG. 1: The main editing page in Creative Labs' SoundFont editor *Vienna* provides numerous parameters for tweaking SoundFonts. Anyone who has ever used a patch editor should find the interface familiar.

have on your computer without requiring additional hardware. Seer Systems (www.seersystems.com) makes two such Windows products: Reality (\$379), a fullfeatured synthesizer with sampling options that supports SoundFonts; and SurReal (\$99), a lower-cost, scaled-back version of Reality. Unity DS-1 (\$449) from BitHeadz (www.bitheadz.com) is another SoundFont-compatible software sampler that runs on Macs and PCs, and NemeSys (www.nemesysmusic.com) has announced that its new GigaStudio software sampler will support SoundFonts as well. DLS software synths include MIDI Wave Lite (\$49) by Galileo Designs (www.galileodesigns.com) and Microsoft's DirectMusic synthesizer, a component of DirectX 6.1 (built into newer Windows operating systems).

Downloadable sounds can also be used in your external hardware sampler. With a converter such as ChickenSystems' *Translator* (www.chickensys.com), you can convert a SoundFont to a format that Kurzweil, Ensoniq, Akai, Roland, and other samplers are able to read. You can also convert sounds from those hardware formats to SoundFonts and perform other types of conversions.

Using SoundFonts in a recording situation has become easy with Sound-Font support from both Cakewalk's *Pro Audio* and Steinberg's *Cubase* digital

audio sequencers. If your sound card is SoundFont-compatible, you simply load a SoundFont bank directly from your sequencer and select your SoundFont instrument in the Track window (see Fig. 2). The process couldn't be easier. Of course, you can play a sound-card synth loaded with downloadable sounds using any sequencer, but not all sequencers support the format as directly as these two.

SoundFont libraries are proliferating as the format gains popularity. Among

the many current producers of Sound-Fonts are Creative Labs, E-mu/Ensoniq (www.emu.com), and SonidoMedia (www.sonidomedia.com). You can also find free SoundFonts on the Web, but their quality is not always guaranteed. Most of the companies mentioned here include product demos on their Web sites so that you can hear what you're getting before you buy.

DLS sound libraries are more difficult to find, but programs such as Awave and Audio Compositor allow you to open a SoundFont and save it as a DLS Level 1 or DLS Level 2 file. Be aware, however, that when you convert from a SoundFont to a DLS Level 1 file, you lose any filtering or layering programmed into the SoundFont patch.

#### **SOUNDFONTS AND DLS FILES**

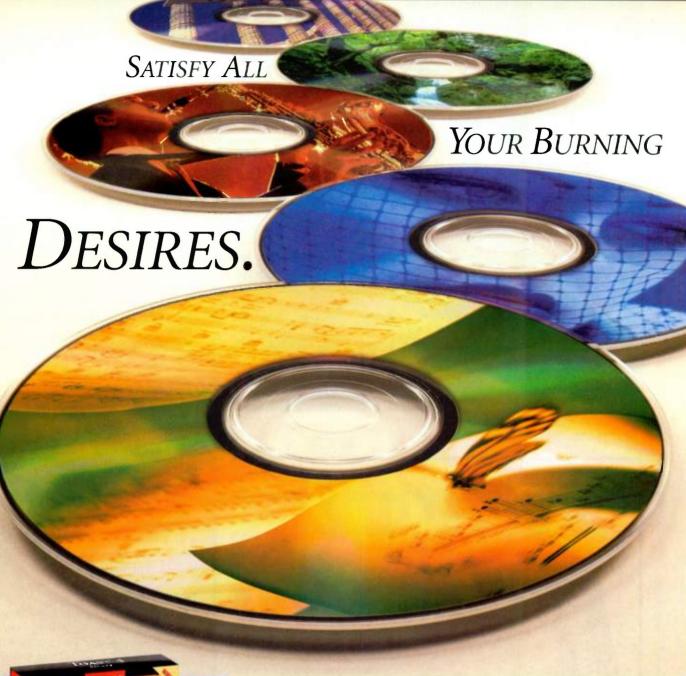
Creating music using SoundFonts and DLS is just like creating music with any type of sound hardware: all you really need is a MIDI controller or a sequencer and the hardware (or software sampler) that plays the selected sounds. Although you can use any sequencer, you'll probably find the task to be faster and easier with a program that can address the sound banks internally.

How you load SoundFonts or DLS sounds depends on the hardware or the software sampler you're working with. Some sound cards have factory- or user-installed RAM into which you can load your sounds, but most use your computer's RAM to store the sample data when you access it. The more system



FIG. 2: SoundFont support is beginning to show up in modern sequencers. Here, the SoundFont bank manager appears inside Cakewalk's *Pro Audio*.





TOAST 4

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FIG. 3: Most sound cards are bundled with software for managing downloadable sounds. The E-mu APS SoundFont Bank Manager, shown here, lets you select a sound bank and preview the Presets that it contains.

RAM you have, the more downloadable sounds you can load at one time. When you're not sampling, the system RAM is available for other applications. (Some cards load sounds into RAM at startup. Check with the manufacturer to see which method your card uses.)

In Fig. 3, we see the screen from which you select the SoundFont file that loads into the E-mu APS card when you start your computer. Simply click the Browse button to locate all of the sounds on your system, then choose the one that you want. You can preview the sounds in the bank using the onscreen keyboard and modify other card settings as well. Loading downloadable sounds is relatively simple; the product manual should explain it all.

#### **FONT OF WISDOM**

As the industry takes hold of downloadable formats, we will see them become even better integrated with audio software. Some of the leading MIDI sequencer developers are already working toward routing plug-in effects onto SoundFont and DLS tracks. Keyboard synth manufacturers are discussing direct support for these formats because of their growing popularity and available libraries. If you're seeking new sonic resources for your music productions, be sure to check out downloadable sounds.

Jennifor Hruska is president of Sonic Network, a network of PC-audio Web sites, including Sonic Implants, the maker of downloadable SoundFont, DLS, Akai, and Kurzweil sound libraries. You can hear her work at www.sonicimplants.com.

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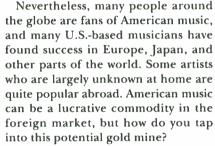


## Taking Your Music Abroad

#### How to make your mark in foreign countries.

By Ken Stockwell

or many American musicians and independent record labels, the large amount of work necessary to market and promote their records domestically leaves them with few resources to pursue international markets and album sales. Marketing and distributing music abroad is a complex process, and legal issues, language barriers, and a lack of knowledge about international radio and retail outlets can hinder even the most ambitious artists.



For the past six years I've helped run Silly Bird Records, an independent music label that has gained substantial international attention, primarily in the United Kingdom and Eastern Europe. In addition to drawing on my own experience to write this article, I spoke with several other professionals about how they overcame international barriers to find success overseas.

One was House, a musician and producer who has toured and released albums in Japan with the bands Limbomaniacs and MCM & the Monster. House is currently working on a hiphop/dub/electronica project called Ben Wa. Another, Dren McDonald, is the owner and operator of Ralph America and Vaccination Records, two independent experimental-music labels that are based in the United States but have achieved success in Germany, the Netherlands, and elsewhere (Ralph America features the widely acclaimed avant-gardists the Residents). McDonald recently launched Clamazon.com, a



#### Roast, Toast, Grin, and Boast

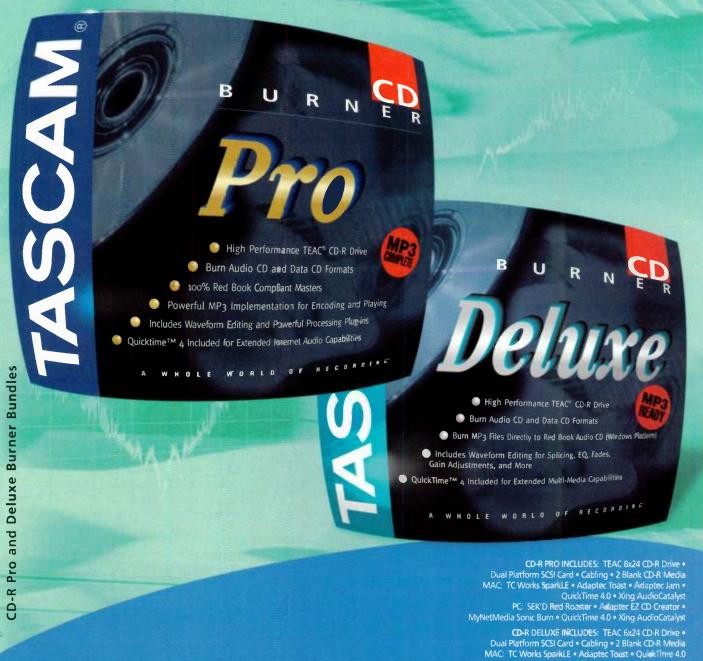
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#### PLANNING YOUR INVASION

Although having your music played on the radio in Sri Lanka or Uzbekistan might sound like a cool idea, don't spread your international campaign too thin. Be sure to target the higher-profile areas first. For most types of music, this means focusing on financial centers in Western Europe (England, France, Germany, Austria, and the Netherlands); Scandinavia

(almost exclusively the cities of Oslo and Stockholm); and Japan. Most international consumers of American music reside in these locations. Concentrate first on these areas, which will be more than enough of a challenge, before you consider extending your campaign into less accessible parts of the world.

Certain regional audiences are more receptive to particular genres than others. It is crucial to research and have an understanding of the music markets in the countries that you want to break into. You may already have some idea of which types of music are popular in other countries. For example, if you're a fan of German techno or British dance pop and your music reflects that influence, you know which places to target. Enclaves of country-and-western or new-age enthusiasts, however, may be more challenging to find. In such cases, you will have to locate your audience before sending out material.

Although releasing your music abroad can be rewarding, it is a difficult, complex, and frustrating process. Unknown American artists who find instant success overseas are not unheard of, but you'll probably have to lay the groundwork for your international music career right here in the States. Radio stations, promoters, and distributors in Western Europe, Japan, and the other primary-target regions will first gauge how well an artist or a group has fared



Don't spend the bulk of your resources trying to get your music played on foreign radio.

on home turf before they will consider establishing a working relationship.

"It's the chicken-or-egg dilemma," says House. "It's difficult to make contacts, set up a licensing deal, and find distribution for your music without a track record. You have to build nameawareness in the United States before you can even consider being picked up for distribution anywhere else—yet you need all of this to start building a track record."

#### **MARKETING DIFFERENCES**

Marketing your music in other parts of the world is similar to marketing it domestically. International music-buyers seeking American artists are fairly savvy and accustomed to the marketing techniques used in the United States. Reproducing promotional materials such as posters and one-sheets in different languages is often unnecessary because English is fairly well known in



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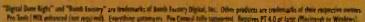
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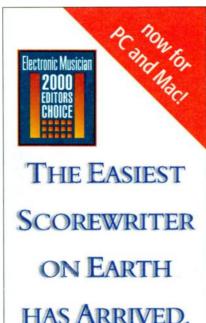


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#### WORKING MUSICIAN

most of the countries you'll be targeting. In fact, marketing in English can benefit your foreign effort—especially in Japan—by adding an "exotic" American flavor to your venture.

Radio is as important abroad as it is in the United States. Trying to get radio play on European or Japanese stations, however, can be a daunting task. Don't spend the bulk of your resources on this effort. "Our German distributor sends a few releases to European radio," says McDonald, "but on the whole, radio stations ignore new music by American independent artists." Silly Bird's success on European radio came not from blind mailings to program managers but rather from relationships with individual DJs that were established through friends and contacts.

Though some small (usually pirate) radio stations may air unknown artists, there is no foreign equivalent to the American network of college and independently owned commercial radio stations. Domestically, Silly Bird sends out more promotional packages to radio stations than to print media, but it is best to concentrate your efforts on print when you're trying to promote your

music abroad. Getting your material reviewed in a magazine is the most effective tool you have when marketing your music internationally. "If you can't get good press, you can't do well in other countries," says McDonald.

House sees radio play as the last step in a long process that begins with obtaining licensing and distribution deals and continues with touring and finding a foreign label to release your music. Only after you have taken these steps will your hit song have a good chance of being picked up by international radio markets.

#### **DISTRIBUTION AND LICENSING**

Distributing compact discs and other media to international markets is rarely cost-effective if you attempt to export directly from the United States. Legal issues surround the sale of foreign products in almost every country, and you can expect to pay 20 to 40 percent in tariffs if you lawfully import your products. "When we sold directly to Germany, we had to mark down the price of our releases to make them affordable in the stores," says McDonald. "It became hardly worth it."

During the PopCom convention in

to music distribution for artists and others in the business.

#### **Internet Resources** A London-based MP3 Web site. **Cerberus Digital Jukebox** cdj.co.uk/cdj International www.medialawyer.com This site offers articles, sample **Entertainment, Multimedia** contracts, advice, and links. The primary focus is on film and & Intellectual Property Law and Business Network audio-visual multimedia, but you will also find advice and information relevant to musicians. IUMA Another source for free MP3 www.iuma.com (Internet Underground downloads from thousands of Music Archive) independent and unsigned artists. www.live365.com Anyone can become a live MP3 Live365.com DJ at this site. Many members pepper their playlists with popular songs to attract listeners in an attempt to get their own music heard. Place your audio files on this MP3.com www.mp3.com site and start getting the word out. Museek.net www.museek.net This site, run by industry experts, aims to help market and promote independent and unsigned acts around the world. MusicDistribution.com www.musicdistribution.net Links and information pertaining



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Cologne, Germany, McDonald kept hearing the same thing: "While a lot of distributors liked us and liked what we did, I found out that they didn't want—or simply couldn't afford—to deal with the tariffs, shipping fees, and other costs. It was just too hard to make the effort worthwhile."

Add to these legal and logistical hurdles the problem of being a relatively unknown entity, and you're in a tough situation. Many small labels and artists form coalitions to overcome these obstacles. Label groups that have affiliations in particular countries, for instance, can avoid many of the tax issues involved in the international distribution of music. If you can make a good case that your partners in France, say, are a part of your business, you will probably avoid French import tariffs. "Even if you just have a friend in Europe and want to set him or her up as an employee of your label, I think that would be beneficial," says McDonald.

The "strength in numbers" advantage is also a great incentive for joining

or creating a music coalition. The larger your presence, the more attention you'll receive. This is more important now than ever, as recent years have seen a recession of interest in independent music distribution. House agrees: "In Europe there is a big shift away from dealing with the smaller labels. To get any real success, you need association with a large entity, such as a well-known group of labels. Most European distributors will not even consider you unless they see a connection to some kind of 'longarmed' entity."

Because distributing and exporting their music involves so many headaches, artists and labels may want to seek out licensing deals with local labels in a particular region or country. These can be one-time deals based on

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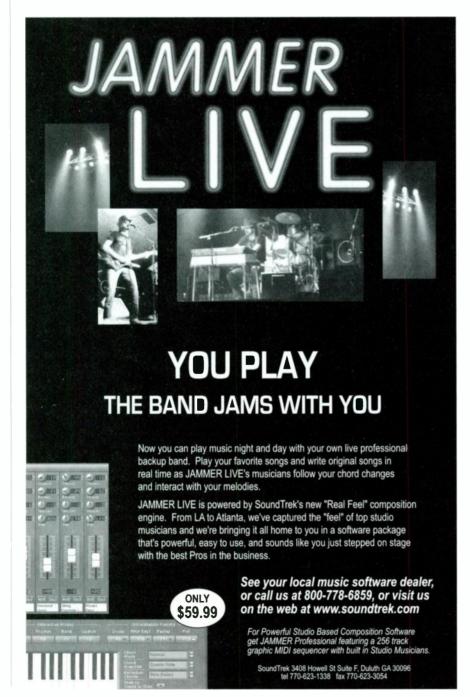
The Internet can help relatively unknown artists find an audience outside the United States.

a sole release, or long-term licensing relationships. The last Limbomaniacs release in Japan was a compilation of songs from the group's first two U.S. albums licensed to a Japanese record label. Vaccination Records has forged a long-term relationship with Flight 13, a German label and distributor that handles most of Vaccination's European releases.

Of course, the trick is to find a deal. When I asked House and McDonald about what worked for them, they both credited a combination of hard work, a little luck, and the use of existing contacts to develop new connections. Creating a relationship with any company takes time, and establishing a successful one with a foreign company can take months or years.

#### MAIL ORDER AND THE WEB

The Internet has created a number of new opportunities for access to an international audience. Hundreds of sites let artists post their music on the Web for download and mail-order



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Bromley, England, 1899\*



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\*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.\*\*



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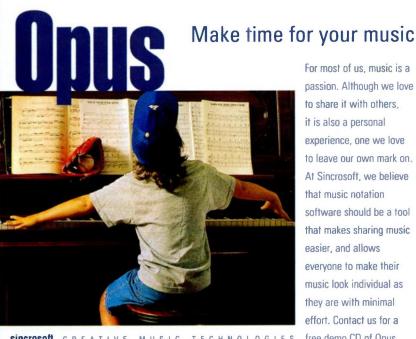
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purposes. For anyone trying to attract foreign attention, the Internet is the best place to start. Putting your audio files on one of the major download or streaming sites—such as MP3.com, IUMA (Internet Underground Music Archive), or Live365.com—is the first step (see the table "Internet Resources"). The next step is to locate a Web site that will take your CDs and arrange mail-order sales.

The Internet can help relatively unknown artists find an audience outside the United States without the work involved in securing foreign licensing and distribution. "We started Clamazon.com to get around the international distributor thing," says McDonald. "The Web has enabled us to get records to people who would never be able to find them in their local stores and might never even hear about them. It's really starting to help us get around this problem."

Nonetheless, attempts to garner international attention for your music will almost certainly fail if you limit yourself to online promotion alone. Even though the Internet can be effective in reaching listeners around the world, it is just one tool in your marketing and distribution arsenal. There is no substitute for the hard work needed to break into foreign music markets.

#### **START PACKING**

No matter which tactics you use to get your music heard abroad, you'll have to push the limits of your abilities and resources. Both House and McDonald stress that the process of working with existing contacts, making new connections, and developing international relationships is more important than anything else. If you don't have the contacts or the ability to travel to target locations, you have that much more work ahead of you.

Before you begin your international campaign, decide where your best bets lie and determine what level of commitment you can put into your efforts. You may need to scale back your goals and focus on fewer locations at first. Embarking on this global adventure will most assuredly be hard work, but you may reap unimaginable rewards.

In addition to running Silly Bird Records, Ken Stockwell is a writer, musician, and broadband-media music producer. He lives in New York City.

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# REVIEWS

Doland VM 2100D.

Native Instruments Transformator 1.0 (Mac/Win)

124 TC Electronic M-One and D-Two

Cakewalk Pro Audio 9.0 (Win)

Lexicon MPX 500

BitHeadz Voodoo 1.1 (Mac/Win)

Waves Native Power Pack II
(Mac/Win)

Kurzweil ExpressionMate

Quick Picks: AIPL WarmTone 2.1
(Win); Cyberwave EMS Waveplant
(Mac/Win); NemeSys GigaHarp
and Upright Acoustic Bass (GigaSampler); GrooveAdemics Jazz Rock
Inversion; Big Fish Audio Roots of
India; Antares Auto-Tune 1.6 (Mac/Win)

### ROLAND

VM-3100PRO

An affordable but powerful digital mixer.

By Rob Shirak

f you are thinking, "Just what the world needs: another digital mixer," maybe you are right—minus the sarcasm. Roland has a fine track record of delivering low-cost, high-quality products. That's exactly what the mixer market needs right now, and the VM-3100 series is a darned good start. Just slightly larger than the average hardcover book, Roland's VM-

3100 provides analog and digital I/O, high-quality multi-effects processing, scene automation, and external MIDI device control, at a price comparable to those of many compact analog mixers.

For those of you who need still more, the VM-3100Pro version adds a second multi-effects processor, speaker-modeling algorithms (for use with Roland's DS-90 digital reference monitors), and a digital bus (R-BUS) for passing eight channels of audio to and from Lightpipe- or TDIF-compatible devices. I reviewed the VM-3100Pro with a DIF-AT expansion box, but for the purposes of this review I will refer to both units as VM-3100 when features are identical. Any detail specific to the VM-3100Pro will be properly noted.

#### **GAZINTAS AND GAZOUTAS**

The VM-3100 processes signals internally at 24-bit resolution; the A/D converters on channels 1 through 8 are 24-bit, and those on channels 9 through 12 are 20-bit. Input channels 1 and 2 have balanced ½-inch TRS inputs, as well as XLR inputs that can be phantom powered; channels 3 through 8 have unbalanced ½-inch inputs. There are gain pots on channels 1 through 8 labeled simply "Line" and "Mic" at the extreme ends of the range. Channel 4



The Roland VM-3100 is a 24-bit digital mixer that offers flexible analog and digital 1/0, internal snapshot animation, MIDI automation, 3-band parametric EQ, and onboard digital effects at a budget price.

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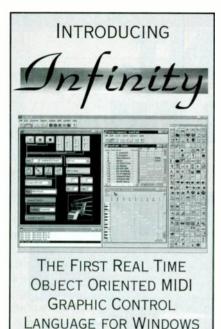
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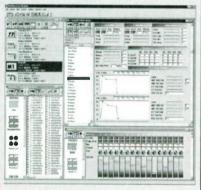


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while the mid band is a peaking type with a variable bandwidth. All bands can be boosted or cut by 12 dB.

If you wish to equalize the Master outs, Bus outs, or Aux Sends, you'll have to patch in an analog EQ, as no internal EQs are dedicated to these outputs and there are no insert points. I would have liked at least a single internal stereo EQ dedicated to the Master outs or, better yet, one that could be assigned to either the Master outs, Bus outs, or Aux Sends. The same goes for the internal effects modules: there are no EQs for the returns. They aren't critical but would have been a nice touch.

Nonetheless, the EQ in the VM-3100 sounds very good, and it is similar to that found in Roland's venerable VS-series recorders. Although not surgically precise, the internal EQs do an excellent job with corrective equalization and creative filtering. Digital EQ gets better every year, and the company has not skimped on the quality of this mixer's EQ, or its effects.

A library holds 16 preset EQ curves, as well as slots for an additional 16 settings that you can name. The library patches can be edited, copied, and deleted. If the channel EQs don't provide enough filtering, you can turn to the more elaborate equalizers in the multi-effects processors.

#### **FEELING THE EFFECTS**

Roland has really improved its multieffects processors in the last few years, and the VM-3100's designers held back nothing. In fact, they pulled many of the mixer's algorithms directly from Roland's latest line of effects boxes. The VM-3100 contains a single stereo effects processor; the VM-3100Pro comes with a second processor that includes some of Roland's COSM-based modeling effects for amplifiers and speakers.

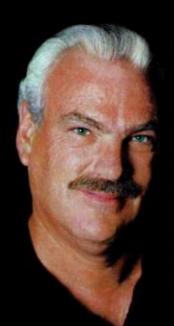
The effects sections of the VM-3100 and VM-3100Pro are quite different and mark an important distinction between the two versions. The single effects processor in the standard

#### VM-3100/VM-3100Pro Specifications

Input Channels (VM-3100/VM-3100Pro) 12/20 Channel EQ 3-band parametric **Dynamics Processors** (2) global Effects Processors (VM-3100/VM-3100Pro) 1/2 (global) Effects Presets (VM-3100/VM-3100Pro) 50/100 **Analog Inputs** (2) balanced 1/4" TRS line level; (6) unbalanced 1/4"; (2) balanced XLR mic level; (1) unbalanced RCA line level; (1) unbalanced 1/4" quitar level **Analog Outputs** (2) unbalanced ¼" Master; (2) unbalanced ¼" Aux Send; (2) unbalanced RCA Bus; (1) 1/2" stereo headphone Digital I/O (1 pr.) S/PDIF optical; (1 pr.) S/PDIF coaxial Other Ports MIDI In, Out/Thru; (1) 1/2" footswitch; (1) DB-25 RMDB II bus connector (VM-3100Pro only) A/D Converters inputs 1-8: 24-bit, 64× oversampling; inputs 9-12: 20-bit, 64× oversampling **D/A Converters** 24-bit, 128× oversampling Sample Rate 44.1 kHz, fixed Display  $136 \times 32$ -pixel (7 segments  $\times$  25 characters) backlit graphic LCD Options DIF-AT interface; ADA-7000 breakout box Frequency Response 20 Hz-20 kHz **Total Harmonic Distortion** 0.002% or less Noise -84 dBu or less Power internal 117/230/240 VAC **Dimensions** 11.81'' (W)  $\times 3.81''$  (H)  $\times 13.56''$  (D) Weight 7 94 lhs

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model allows you to store edited effects only in scene memories, not in user locations. The Pro version, on the other hand, has a Module page where you can turn certain effects on and off, and a Parameters page where you can edit effects. For the additional \$300 that the VM-3100Pro costs, you get not only a second effects processor but the ability to edit both internal processors.

The VM-3100Pro's multitude of effects, as well as the parameters available for most of them, is overwhelm-

ing. The only problem with the effects processors is that there aren't enough! The reverbs, delays, and choruses are excellent all around, and the COSM-based speaker modeling and guitar-based effects are very well done. The combination of guitar input, multi-effects, and amp simulation provides a very capable guitar rig for direct recording. You don't need anything else except a guitar, a cord, and a performer. I own a Boss GT-5 guitar multi-effects processor, and the VM-3100's effects compare with it admirably in

both capabilities and sonic quality.

As with the EQ library, you can store effects as patches for later recall. In addition to 99 presets, you get 99 user slots for storing edited patches (VM-3100Pro only).

The mixer also has two dedicated compressors that can be assigned to any channel except the Master. I didn't find them particularly inspiring, but they were functional. Because only two compressors are available, it's a good idea to use outboard compressors as well while recording. On the other hand, you'll find additional compressors and limiters in the effects section.

#### **PLAYS WELL WITH OTHERS**

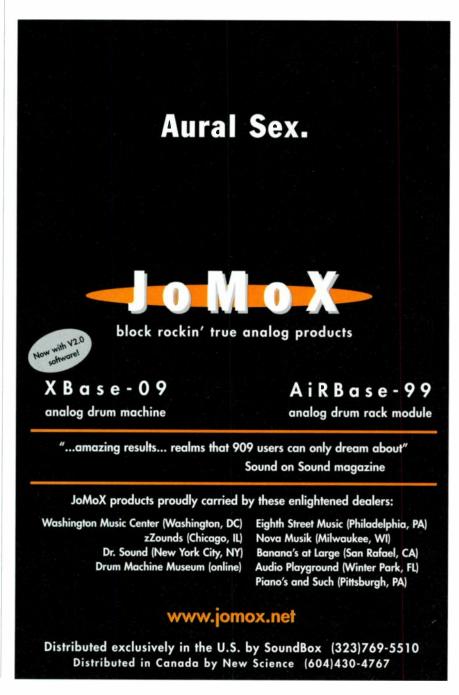
The DIF-AT option (VM-3100Pro only) allows the transfer of eight tracks of digital audio between the VM-3100Pro and Alesis ADAT-compatible or Tascam

When it comes
to sonic quality,
the VM-3100 is a
real winner.

DA-series devices. The DIF-AT interface contains sync and audio I/O connectors for both ADAT and Tascam devices in a single box, eliminating the need for different format interfaces. Since synchronization is provided for both formats, you can control the transports of both devices by using the VM-3100Pro's transport controls.

A 25-pin RMDB II cable connects the VM-3100Pro to the DIF-AT. (The company stresses that only cable manufactured by Roland should be used, to prevent damage to the equipment.) Toslink (optical) connectors on the DIF-AT are for ADAT audio I/O, and a TDIF-1 port transfers tracks to and from a DA-series recorder.

The VM-3100Pro's major drawback is its fixed 44.1 kHz sampling rate. This means that any MDM tapes formatted at 48 kHz (ADAT or Tascam) will not play back at the proper speed when connected to a VM-3100Pro through the DIF-AT. With most current ADAT and DA-series recorders,



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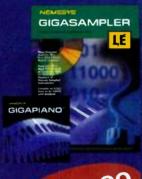
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## **NEMESYS**

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you can eliminate the problem simply by formatting future tapes to 44.1 kHz and maintaining this sample rate. If you've used 44.1 kHz for past projects, everything will work fine. The problem surfaces only with older tapes already formatted and recorded at 48 kHz, and with all recordings made with older blackface ADATs, which operate only at 48 kHz.

In these instances, Roland recommends connecting the recorders to the VM-3100Pro via analog connections. This will surely work, but it misses the point of using a digital mixer. Here's a work-around for blackface ADAT users: either make new recordings to the ADAT while synched to the VM-3100Pro (which will slow down the transport to 44.1 kHz), or pitch the blackface ADAT down 147 cents when recording independently of the DIF-AT/ VM-3100Pro setup. This will effectively change the sample rate from 48 to 44.1 kHz. The ABS time readout will then be slower than real time, but the VM-3100Pro's time-code readout will be correct.

Sampling rates aside, the DIF-AT worked right out of the box. I successfully established digital I/O and sync

**ROLAND** 

VM-3100 digital mixer

\$995

connections between the VM-3100Pro and my blackface ADAT. I didn't get the chance to test a DA-88, but I haven't heard any reports of problems with either data transfers or synchronization. If you have an ADAT-compatible or DA-series recorder or sound card and are interested in the VM-3100, remember that you must purchase the Pro version.

SLOW FADE

The VM-3100 offers a lot of features for the hobbyist, semipro, or professional with a small MIDI rig or recording setup. The quality of the preamps, A/D/A converters, EQ, and effects is excellent on both models, and the two effects processors on the VM-3100Pro allow for an extraordinary variety of editable parameters. The fact that you can store mixes to scene memories and manually switch them adds to the VM-3100's usefulness. And don't forget that the unit includes a whole guitar rig to boot.

Is the VM-3100 the right mixer for you? That depends on your specific needs. It has a lot of great features but is missing a few things that are important in certain studio environments. I wish it had a dedicated monitoring section, EQ, and compression for the Master and Bus outs, and individually definable crossfades between scenes. Also, the small number of analog inputs may make the mixer unsuitable for MIDI setups that contain a lot of synthesizers and modules. However, you can expand the VM-3100's analog input capacity by using the optional ADA-7000 breakout box (\$1,245), which adds eight XLR and 1/2-inch TRS balanced inputs. You could also use an outboard patch bay.

Limitations notwithstanding, when it comes to sonic quality the VM-3100 is a real winner. And if you have an ADAT or DA-series recorder or sound card, the DIF-AT option is a must for providing eight more channels of I/O. Just remember that if you go for the DIF-AT, you'll need the fully featured VM-3100Pro mixer.

Producer/songwriter Rob Shirak (formerly Shrock) has elected to change his last name back to its traditional spelling. This has made his parents very happy, but his wife—undergoing a temporary identity crisis—is going by her maiden name until further notice.



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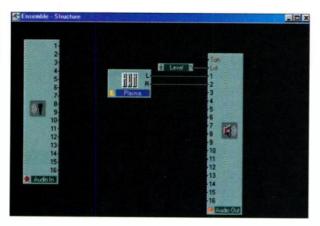


FIG. 2: The Ensemble is the highest level in a modular network's hierarchy. Even though an Ensemble can look relatively simple, its elements summarize other networks below its level.

right-click on the Plasma box in the Ensemble window and select Structure. The details of Plasma's modular network will then be more apparent (see Fig. 3). For example, the leftmost box, Notein, is the beginning of Plasma's Instrument structure. To find out which functions Notein performs, right-click on the box and select Properties from the pop-up menu; a dialog box will appear and describe the details of what Notein does. You can access the same information by positioning the mouse pointer over the Notein box; the description then appears in a highlighted window. This works with any input or output.

The Notein box is itself a summarized view of an even lower level in Plasma's structure, the Macro (see Fig. 4). A Macro can be thought of as a structure within a structure, except that it doesn't manage MIDI data or have control panels. A Macro is identifiable by its gray label and "structure" icon. As you can see in Fig. 3, the boxes labeled Pitch, Position, Re-Synthesis, Amp, Filter, and Diffusion are all Macros. Each contains a lower network level.

The most basic part of a network's structure is the module. Like Generator, Transformator contains a variety of modules that perform basic tasks such as audio and event switching, MIDI event generation, mathematical functions, audio mixing and amplification, and sequencing. As you can see in Fig. 4, this network of modules delivers MIDI note values for pitch and gate (note on/off) to the Pitch and Amp Macros shown in Fig. 3. The Macros then connect to the Sample/Map module, which is actually the Sample Resynth module (more details in a moment).

#### **ART OF NOISE**

Transformator employs seven types of Sampler modules to play back sound samples. The most basic module is simply called Sampler. It has a logarithmic control input for playback rate (pitch) and for selecting a sample from the loaded sample map. It also has a trigger input that plays back the sample from the beginning, a control input for setting the output amplitude, and an output connec-

tion (on the module's right side).

Another relatively simple module is Sample Lookup. It makes samples available as function value look-up tables. You set a position within the sample with the Position input, and the value is sent to the outputs. This allows you to play back samples from a point other than the sample's beginning. Sample Lookup adds separate Left and Right audio outputs so that you can play stereo samples.

The remaining Sampler modules become increasingly complex. The Sampler FM module adds a linear control input for modulating the sample playback rate, along with a control input that determines the starting point when the next trigger occurs. The output side adds a polyphonic event output that dictates sample playback length. The Sampler Loop module adds a gate input control, a control input for selecting a sample from the sample map,

and a Loop Start and Loop Length input control.

The Sample Resynth module is a realtime resynthesizer that allows you to control pitch and playback speed independently and manipulate samples extensively. Standard samplers maintain a pointer for each voice that always points to the current position in the sample. Output amplitude is always the same as the amplitude of the sample at the current pointer position. Because the pointer moves at varying

You can create
or reproduce
instruments
with nonstandard
tunings.

speeds through the sample, it generates an amplitude that varies over time. In other words, the pointer's movement speed determines sample playback speed and the pitch at which the sample is heard. If the pointer stops moving, output amplitude stops changing—which means that no audible signal is produced.

With Sample Resynth, a synthesizer inside the module resynthesizes the signal at the pointer position to generate the output signal. The pitch of the resynthesized signal is independent of the sample's pointer speed.

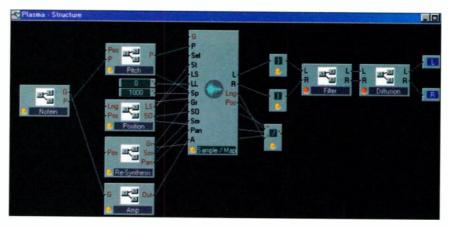


FIG. 3: The Instrument level falls directly below Ensemble in the modular network's hierarchy. Here you can see Ensemble's makeup in greater detail. Note that many of the elements shown on this screen are Macros, the next level down in the hierarchy.



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#### **FUSION REAKTOR**

Although Transformator and Generator are available as separate programs, you may want to consider buying them in one package, marketed by Native Instruments under the name Reaktor. For one thing, Reaktor costs just \$469, while Generator and Transformator are \$298 each. But buying Reaktor is more than a matter of saving \$127. It provides a unified environment in which you can use Generator and Transformator together. Imagine, for example, creating a multitimbral instrument that performs FM synthesis and uses analog waveforms, multisampling, and granular synthesis simultaneously. With Generator's FM and subtractive synthesis features and Transformator's sampling and granular synthesis capabilities, you can do all of that and more. But you can't achieve the same results using both programs separately.

Reaktor's copy-protection scheme loads a 100 MB file named Enigma onto your computer's hard disk; it acts as the "key" for program operation. I'm glad that every software maker doesn't use this sort of copy protection; if they did, keys would occupy most of my drive space. On the positive side, at least Native Instruments

didn't use a hardware dongle or require a CD-ROM to be loaded at all times.

Another reason to upgrade is that Reaktor has new sound-creation modules such as Multisine and Geiger; mathematical module tools like Sine/Cosine, Square Root, and Modulo; and routing modules including To Voice, From Voice, and Router 16. Reaktor also contains additional Macros and Instruments to make the process of creating these elements less tedious.

Because the program supports ASIO 2.0, so you can route different instruments in a multitimbral Ensemble to separate outputs, provided you have a multichannel audio card with ASIO drivers. Furthermore, *Reaktor* can be a VST 2.0 plug-in: you can open an Ensemble as a virtual instrument, or use any of the package's effects (*Reaktor* considers effects processors to be Instruments) to process audio tracks in Steinberg's *Cubase VST* or a VST 2.0-compatible sequencer such as Emagic's *Logic Audio*.

For Mac users, Reaktor also supports DirectConnect, which provides an easy way for applications to stream audio directly into Pro Tools or other DAE hosts. DirectConnect allows up to

32 separate audio channel outputs from any host-based application (such as software synthesizers or samplers) to be independently routed, recorded, processed, and mixed within the Pro Tools TDM mixing environment. Currently, Reaktor supports a maximum of 16 DirectConnect audio channels. Routing audio input into Reaktor using DirectConnect isn't yet supported by Digidesign, but it's in the works.

In addition, Reaktor includes MIDI File Player, so you can now play Standard MIDI Files without opening a separate sequencer. This reduces the amount of system resources your computer needs to play back Reaktor Ensembles or Instruments, making system overload less likely and thus helping to prevent the skips and dropouts caused by resource allocation problems during operation.

Overall, Reaktor is more efficient than the separate Transformator and Generator programs in terms of the average percentage and peak use of resources. With a software-based synth or sampler—especially a modular program—efficiency is a major concern, and it's one that Reaktor addresses admirably.

up your sound sample files with the Sampler module. And once you have created your Sample Map, you can save it as a separate file. When you want to use a different Sample Map with the same Sampler module, all you need to do is load another Sample Map file—all settings and sounds will be loaded into the appropriate Sampler module.

However, once sound files are loaded into a Macro, Instrument, or Ensemble mode, *Transformator* saves only the paths of the associated sound files. If these files are deleted, moved, or renamed after the structures are saved, the program won't be able to find the files—which makes it nearly impossible to re-create the structures in question.

Nevertheless, there are two ways around this problem. First, *Transformator* opens a dialog box to indicate which sample files are "missing." Using

the Replace button, you may be able to find and reload the moved or renamed files. The alternative is to back up the sound files within the Macro, Instrument, or Ensemble file as mentioned in the previous paragraph. This way, if something happens to the original files, you can use the copies to restore your work.

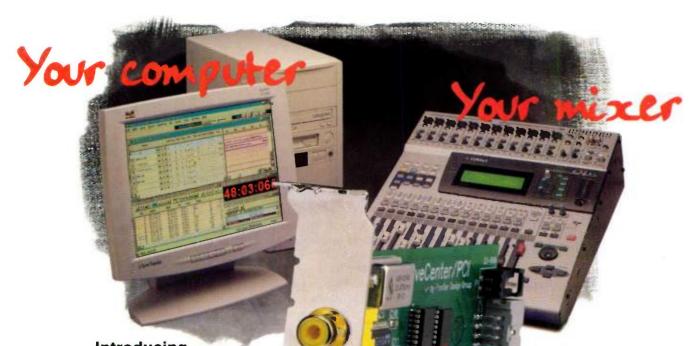
#### **ANALYZE THIS**

The biggest problem I faced while reviewing Transformator was grasping how useful modular sampling can be. I suspect that this lack of vision will be a significant barrier for other users as well. Transformator did a creditable job of enlightening me, however, by providing good examples of the different uses for modular sampling. I was pleasantly surprised to discover that modular sampling can be as simple or as complex as you need or want it to be.

Even so, for conventional software sampler tasks, I prefer a program that better mimics a standard sampler without all of the hassles inherent in a modular-based program. Call me old-fashioned, but I'm accustomed to that traditional interface.

On the other hand, when it comes to playing synchronized audio loops and sequenced percussion samples, or for combining sample playback with synthesized sounds, *Transformator* really shines, especially when it works in the *Reaktor* environment (see the sidebar "Fusion Reaktor"). And when you use its granular synthesis tools to mangle sounds beyond recognition, *Transformator* is pure, unadulterated aural fun. Plug into it and catch a good buzz.

**Zack Price** occasionally claims to be a former member of a famous German synth band, just to see if people will really believe him.



Introducing WaveCenter/PCI

OK, you see what's happening: digital mixers are looking pretty cool. After all, they've got incredible sonics, built-in effects, and the automation capabilities you could only dream about before. But if you hook that puppy up to the NoiseRacket analog soundcard that came with your computer, you're right back in \* \* \* \* ville. (Rhymes with "Snapville.")

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# TC ELECTRONIC

M-ONE AND D-TWO

TC comes out swinging in the arena of budget multi-effects processors.

#### By Mike Collins

ooking for a powerful and high-quality reverb or delay unit for less than \$700? Check out the new M-One and D-Two processors from Denmark's TC Electronic. TC introduced several new effects units last year, including the M3000 and Fireworx, but all cost well over \$1,000. This year the company has shifted focus with the introduction of two lower-cost units, each priced at \$699.

The M-One dual effects processor offers some of TC Electronic's most coveted algorithms, including enhanced reverbs based on new research and a good selection of other popular effects. More than 20 different algorithms, with 100 factory presets and 100 user locations, are accessible through the fast, simple, and intuitive user interface.

With up to ten seconds of delay, the D-Two provides six direct-access effects, with 50 presets and 100 user locations. And based on the classic TC Electronic 2290 delay, its multitap rhythm delay offers a very musically oriented Rhythm Tap feature.

#### **GUTS AND GLORY**

For the price, both units have very impressive technical specifications, with 24-bit A/D/A converters; 24-bit internal processing; lots of headroom; low distortion; a wide, flat frequency response;

and a dynamic range rated at better than 93 dB. In addition, you can control just about every parameter through SysEx or continuous controllers.

The units have identical rear panels that feature electronically balanced stereo inputs and outputs on %-inch jacks and S/PDIF I/O on RCA jacks (see Fig. 1), at up to 24-bit resolution with either 44.1 or 48 kHz sample rates. Both also offer a %-inch Pedal Input jack and MIDI In, Out, and Thru ports.

#### **PANEL OF EXPERTS**

Let's look at the M-One processor first. The left side of the unit's front panel sports a power button and three knobs—In Level, Mix (between dry and wet signals), and Effect Bal (the balance between the outputs of the two effects engines).

The center of the panel features a large LCD that includes input meters (with overload indicators), an analog/ digital input-selection display, and a sample-rate readout. A Routing indicator displays the signal flow of the selected routing mode, and Algorithm displays the type of effect used for each engine. Two meters show gain reduction for the dynamics algorithms. The preset number and type appear on the display, which includes a light that tells you whether the current preset has been modified. You can also see the location of the current patch (Factory or User bank). A MIDI In icon indicates the presence of incoming MIDI data.

To the right of the display are three groups of buttons—Setup, Effects, and Program. Beside them is the Control section, which contains a large dataentry wheel surrounded by four more buttons—Enter, Exit, and up and down arrows.

In the Setup section, the Routing key selects the routing mode; the I/O button lets you choose between the analog and digital inputs, set the sample rate, and more; the Tap key allows you to tap a tempo or enter the tap menu; and the Utility key provides access to MIDI, SysEx, Routing Lock, Bypass, and Pedal functions.

The Effects group has four buttons: Algo/Edit 1, Algo/Edit 2, Bypass 1, and Bypass 2. Algo/Edit 1 and 2 let you select and change the algorithms in each effects engine. With Bypass 1 and 2, you can select one of three modes for each engine. Mode 1 passes the input signal directly to the output; the dry signal is set to 100 percent and the effects signal to 0. Mode 2 (Bypass FX Input) cuts the input to the effects engine so that the effect dies out gradually-that is, it rings out for its normal durationrather than abruptly. Mode 3 (Bypass FX Output) zeros the level of the effects output but leaves control of the dry signal's level to the Mix knob.

The Program section has two buttons, Recall and Store. You can use these in conjunction with the Control wheel to select a preset.

Finally, in the Control section, the up and down keys allow you to navigate through the display, and the data wheel lets you change values. Press the Enter key to confirm operations; press Exit to leave a menu or cancel an action.

#### M-ONE FOR THE MONEY

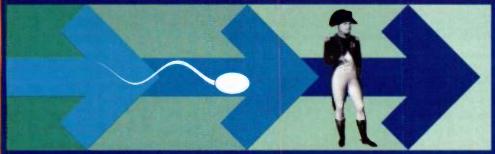
The M-One's user interface is well designed, and I found my way around most of it without having to look at the manual. I did need to look up the Tap function and the Routings, so let's begin with those.

The Tap function lets you tap out a tempo for the Delay time, Chorus rate, and so forth: just press the Tap key once, and the Tap delay parameter appears at the bottom of the display. Tapping on the key in time with the music sets the delay to the approximate value; you can then use the Control wheel to fine-tune it. Pressing the down



TC Electronic's M-One (a) and D-Two (b) processors offer easy access to editing parameters, as well as informative front-panel displays.

## Small, but effective





It's no surprise that something so small can accomplish so much. The D16 has all the capabilities of a full digital recording studio because it is one. It can record, mix, master and then burn your CD\*. It can even emulate your favorite mics and amps to deliver a big studio sound in a compact unit. Sometimes smaller is better.

Digital Recording Studio



- · 24/16-bit recording
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- · 8 virtual tracks per track
- Direct guitar input w/expression pedal input
- 215 PCM drum patterns

- Built-in SCSI for recording/backup to external drives
- · Built-in tuner and mic
- \* Direct CDR CDRW burning (CD burner required)

KORG



FIG. 1: The M-One and the D-Two have identical rear panels.

key brings up Tap Subdivision (the rhythmic subdivision you want to tap along with), which you also set by using the Control wheel. Pressing the down key again brings up the Tap Function selector, letting you select which engine the Tap control works with—1, 2, or both. Tapping down once more takes you to MIDI Sync, which you can set to on or off. Adjusting the parameters is completely straightforward: simply use the cursor keys to select parameters, and spin the Control wheel to select values.

#### **RURAL ROUTE**

I found the routings difficult at first. Six are available: Dual Send/Return, Parallel, Serial, Parallel/Serial, Stereo Linked, and Dual Mono.

Dual Send/Return routing allows you to use the unit as two independent effects processors. Parallel routing sums the left and right inputs, feeding both engines with the same signal. This is great when you want to add two different effects to the same source—for example, a chorus and a reverb to the same guitar track—without their interfering with each other.

In Serial routing, the signal passes through engine 1 before it goes through engine 2. Thus, you can combine effects (for example, a de-esser in engine 1 with a bright reverb in engine 2).

Parallel/Serial routing is similar to Dual Send/Return routing with one exception: the output of engine 1 can be fed back into the input of engine 2 (see Fig. 2). Use this routing when you desire separate inputs to each engine but still want the effects to be partially combined. For example, maybe you have a long delay running on engine 1 and a large hall reverb on engine 2, and you want to use both effects on a lead vocal. If the repeats from the delay seem dry compared with the reverb, you can bleed some delay from engine 1 into the reverb in engine 2 by turning up engine 2's crossfeed parameter. Now you'll hear reverb on both the vocal and the delay repeats.

In Stereo Linked routing, both engines produce exactly the same effect with synchronized parameter settings.

This is the routing to use for true stereo operation—for example, when you use the M-One as a stereo compressor on a subgroup.

In Dual Mono routing, the two engines are completely independent, with mono in and mono out for each engine. Use this if you need effects that are frequently applied in mono rather than stereo (a tremolo on one channel and an EQ on another, for instance).

Each preset in the M-One stores its own routing. What if you want to change presets but keep the same routing? No problem. The Routing Lock function keeps the routing intact as you change patches, no matter which routing is saved with the preset. In addition, preset changes are instantaneous and noninterruptive to the sound—very impressive.

#### **ALGORITHM METHOD**

Two defining elements of a reverb are its early reflections and its tail. On the M-One, you can tweak these two elements with Reverb Tail Decay; Predelay; Size, which indicates the size of the reverberant space; and High Cut, which reduces sibilance in the reverb. Another parameter, High Color, adjusts the decay time in the upper frequencies and reduces sibilance. Low Color adjusts the reverb time in the low frequencies to remove rumble while keeping the warmth of the reverb tail. You can also set the Early Reflections level, Reverb Tail level, and overall effects

level. Modulation speed and depth control the two modulation types, Smooth (which does not detune the source) and Vintage (which tends to detune the source slightly).

By juggling these parameters, you can get natural-sounding reverbs using early reflections and complex, nondetuning modulation; or you can achieve more unusual sounds, including a classic re-

verb without any early reflection patterns, heavily detuned modulation, and so on.

The M-One also emulates older-style reverbs such as plate and spring, and its Live reverb algorithm is optimized for P.A. system applications in spaces where typical reverbs don't cut through. While primarily a reverb unit, the M-One also has a full complement of delay, EQ, and dynamics effects including chorus, flanger, phaser, compressor/limiter, de-esser, gate/expander, tremolo, pitch shifter, detune, and a 3-band parametric EQ.

#### **DENSE AND DENSER**

TC Electronic claims that the M-One's reverb algorithms are more advanced and have more density than those of comparably priced processors. Based on what I've heard, I'm inclined to agree. Sonically, the M-One beats any of the reverb plug-ins on my desktop system. The Natural Hall + Ambient patch, for example, added life, depth, and richness to my solo guitar tracks, and Large/Small Chamber sounded just right on a Latin/pop track with an up-front female lead vocal. The delay effects are hot as well. Check out the one called M-One Magic, a very special delay that's sophisticated yet spooky sounding, with subtle repeats.

If the M-One is so great, then why would anyone spend two or three times as much for TC Electronic's flagship

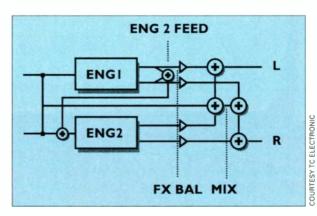


FIG. 2: In Parallel/Serial routing, the output of engine 1 can be fed back into the input of engine 2.

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M2000 or new M3000? Professionals may demand such critical features as the M2000's XLR connectors and AES/EBU digital I/O. But if you're after the exceptional TC reverb algorithms alone, the M-One has the edge. (Keep in mind that for the price of an M2000, you can buy both an M-One and a D-Two.)

The M3000, on the other hand, is definitely a superior unit. It has all the professional features of the M2000, including the new and more natural-sounding VSS reverb algorithms (not included with the M-One), and more than twice the number of presets included with the M-One.

#### **D-TWO FOR THE SHOW**

The main focus of the D-Two is sound quality and operation speed. Based on the classic TC 2290 delay, the D-Two has all the features you could wish for and then some. In addition to providing comprehensive control of tempo parameters, the unit allows you to customize rhythm patterns. For example, you can tap in patterns of up to ten steps directly from the front-panel key (or using a footpedal), then adjust the Pattern's tempo to your taste.

Using the automatic Subdivision feature, you can tap the tempo and the D-Two will automatically adapt to the subdivision that you want. Track Tap is another great feature. When you enable Track Tap, the preset will instantly track the current basic tempo and adapt to it instead of using the

tempo stored with the current preset. This box means business!

#### **FACE VALUE**

The left side of the D-Two's front panel features a power switch, a pair of knobs labeled Input and Mix, and a large LED screen. The display provides a stereo input meter, a delay-time indicator (which registers in user-selectable milliseconds or bpm), an indicator that blinks to show tempo and rhythm, a subdivision indicator, a dynamics meter (which shows gain reduction for the Dynamic Delay algorithm), the preset number, and icons that indicate whether the current preset has been modified and whether you are using the Factory or User bank. As with the M-One, a MIDI In icon shows the presence of incoming MIDI data.

Beneath the delay-time indicator, several others show sample rate, feedback percentage (which determines the decay of the delay repeats), and feedback number (the exact number of repeats). Three LEDs show the Feedback High and Low Cut filter setting, the overall High and Low Cut filter setting, and Ping Pong indicator.

The rest of the front panel is filled with control buttons and a couple of large rotary controls. The first set of buttons includes the Delay/Tap and Feedback/Rhythm keys. Hitting the Delay key once allows you to change delay times, either by using the wheel or by tapping on the Delay key; the D-Two measures the time interval between

the last two taps and calculates the delay time accordingly. The Feedback/ Rhythm key has three main functions. After hitting the key once, you can use the Delay wheel to set the feedback percentage. Pressing and holding the key lets you use the wheel to change the number of repeats. You can also use the key to tap in a rhythm pattern of up to ten steps.

The next section, labeled Effects, allows you to selectively enable or disable six different effects using six dedicated keys: Spatial, Filter, Chorus, Reverse, Dynamic, and Ping Pong. (Pressing these twice jumps you to the relevant edit parameters.) Following that section, four Function keys provide Edit, Setup, Recall, and Store functions. The Setup key opens the setup menu, and the Edit key gives you access to the preset parameters.

Up and down keys to the right of the Function keys allow you to select parameters, and the second data wheel changes their values. Enter and Bypass buttons are located on the far right of the front panel; they function identically to those on the M-One.

#### **DELAYED GRATIFICATION**

A standard delay line produces repeats by feeding the delay output back to the delay input. The D-Two supports this traditional mode of operation, with up to five seconds of delay in stereo and ten seconds in mono, and also provides a couple of other modes that employ a multitap-delay design.

The Straight delay mode lets you control the exact number of repeats—a great feature for tweak heads. The multiple output taps from the delay line produce delays at different points in time. Using the Feedback number function, you can set the number of taps to produce what will sound like simple repeats of the first delay heard. Then, using the Feedback percentage control, you can set the feedback amount for the last tap, which is fed back to the delay input in the multitap modes. This starts the whole process again, so you hear the entire sequence of taps repeating in rhythm.

Be careful not to go over the top, though. The maximum delay time must be shared among the number of taps specified (that is, with ten repeats, you have only one second available on each tap in mono mode). Interestingly, a shuffle parameter lets you add a shuffle feel to the delay repeats in Straight mode.

#### M-One and D-Two Specifications

Analog Inputs	(2) balanced ¼"
DOMESTIC OF STREET	
Analog Outputs	(2) balanced ¼"
Digital I/O	(1 pr.) S/PDIF (RCA)
Other Connections	(1) ¼" footpedal; MIDI In, Out, Thru
Frequency Response	20 Hz-20 kHz
Dynamic Range	104 dB (typical)
Total Harmonic Distortion	< -94 dB (0.002% @ 1 kHz)
Factory Presets (ROM)	M-One: 100; D-Two: 50
User Locations (RAM)	100
Effects Types (M-One)	reverb, delay, chorus, flange, pitch shift, EQ,
	compressor/limiter, gate/expander, de-esser, tremolo, phase
Effects Types (D-Two)	delay, reverse delay, spatial, filter, chorus,
	dynamic (ducking), ping pong
A/D/A Conversion	24-bit
Sampling Rates	44.1 kHz, 48 kHz
Dimensions	1U × 8.2" (D)
Weight	4.1 lbs.

# PLUG-IN SOME REAL MUSICIANS









ommunity hall, and practice. You know, go over a song, or a riff, or an idea, again, and again, and again until you were either completely sick of it, or got it right. Finally you'd just about cobble a set together when the bass player would go off with some girl (normally, yours), or get arrested or something, and you'd have to start the whole grisly process over again.

Sure this was 'paying our dues'. Sure this was fun (sometimes). Sure this was pretty much the only way anyone did anything. But we were only as strong as our weakest member, and thoroughly dependent upon a whole raft of factors other than The Music.

1985 everyone except keyboard players felt under threat from the metronomic march of computers. Would there be such an animal as a drummer in the year 2000? Laughable, now, of course when the Rhythmically Blessed from turntablists to acoustic percussionists practically rule the roost. But computers did stand music on its head for a while. Music became stiff and predictable, and musicians became lazy and unadventurous. Better just to steal a sample than have to wrestle with such concepts as creating a groove of your own. For a while The Music definitely played second fiddle to The Technology.

Here in the 21st century It is Schizold, man. You can mix and match live players with computer control. You can blend MIDI tweakability with audio security. You can buy a complete PC studio for what it used to cost to hire Abbey Road for a day.

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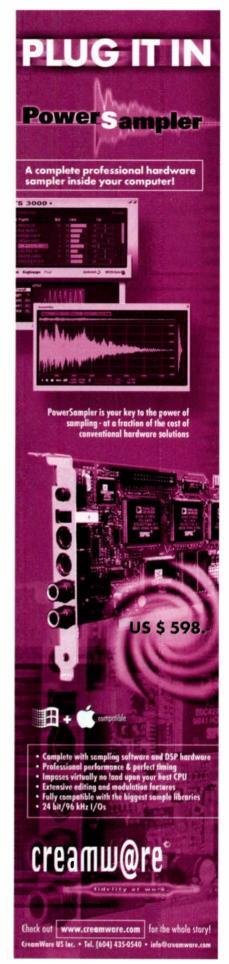
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VOLUMI



Things get even more interesting in Rhythm mode. Here, you can tap in the exact rhythm that you want using the Feedback/Rhythm key and then quantize the pattern to a specific subdivision. You can also edit the pattern and even change the level of individual taps. This much control allows some great customized delay patterns. (As with the Straight delay mode, Rhythm's maximum delay time is divided by the number of specified taps.)

#### QUICK ON THE DRAW

The dedicated Delay & Feedback wheel, the two tap keys, and the six direct-access Effects keys all make the D-Two easy to operate. For example, hitting the Spatial key instantly broadens the delay picture by introducing a small time difference between the two channels. You can also phase-reverse one channel, which widens the delay even more. Hit Ping-Pong, and the delay instantly pans from left to right with five patterns to choose from: hard left to hard right; taps at left, center, and right; and Dynamic. The Dynamic pattern uses as many pan positions as there are delay repeats—for example, a delay with five repeats will have five pan positions from left to right.

The Reverse delay feature is very sophisticated, with a number of styles to choose from. Still, it has limits. For instance, the maximum delay is halved to 2.5 seconds in stereo and 5 seconds

TC ELECTRONIC M-One multi-effects processor \$699 FEATURES EASE OF USE AUDIO QUALITY VALUE 1 2 3 4 5 PROS: Offers flexible modulation of param-

eters via SysEx and continuous controllers. Routing Lock retains desired internal signal flow, regardless of patch changes.

CONS: Doesn't allow modulation of parameters by velocity or note number. Internal processing doesn't provide as much headroom as other, more expensive TC Electronic units.

in mono. If you want to clean up the effect, first hit the Dynamic key and then set a threshold and release time so that the input signal controls the level of delay. When the input signal level goes above threshold, the delay level decreases; when the signal drops below threshold, the delay level increases. Thus, the delays are heard only where you really want them—in the musical

Switching presets is instant, with no interruption to the sound.

pauses—which keeps the sound nice and tidy.

If you want to smooth out the delay sound or simply change its flavor, the Chorus key allows you to add chorus or flange along with the delay. To more closely mimic analog delays, you can apply filters (both in the Delay line and in the Feedback loop) to the repeats to progressively remove more high frequencies as they ring out.

#### **ECHOPLEXED**

Some of the D-Two's echo patches have obvious names, such as Tape Echo, Chorused Delay, Straight 2290 Delay, and My Old Echoplex. Others, such as Shuffle Your Feet and Stabbed in the Back, have less familiar but quite memorable names. Their sounds are memorable, too-once you've heard the effect, you'll know what to look for next time.

Moving Hat gives you instant spacey dub effects, filtered and flanged with an unpredictable rhythm. Low Cut 1/4th Notes is great for reggae, with all of the low frequencies filtered out on the delay repeats. Overall, I liked what I heard and found myself wishing for many more than 50 presets.

#### SYNCHRONIZIN' RHYTHM

The D-Two manual contains lots of useful operational hints and suggested applications. For example, if an engineer in the studio wants to use a multitap delay with a rhythmic relationship between the taps and the lead vocal,

# Throw away your mouse. Throw away your keyboard. And plug in to your areative potential!

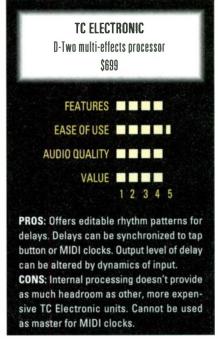


but does not have the tightest timing sense, the D-Two can quantize to the most appropriate subdivision. After you tap the desired rhythmic pattern on the Rhythm Tap key, the multitap Delay is instantly running and in time. To create a fast, jungle-style rhythm delay synchronized to a sequencer, you can tap and edit the pattern at half speed and then select 16th notes. When enabling the MIDI Sync function, you can select the 2:1 setting to create a double-speed feel for the rhythm pattern. On top of this, the D-Two has real-time controls

for the filters, the flanger speed, the enabling and disabling of the Reverse function, and so on—all using MIDI continuous controllers.

When touring, you might want to reproduce, for instance, that very significant three-tap panned Delay used on your band's recording. Just activate the Ping Pong key, limit the repeats to three, and tap in the Delay time. That's it—fast and efficient!

The D-Two does not output MIDI clocks, which would have been a valuable addition to its feature set. Imagine



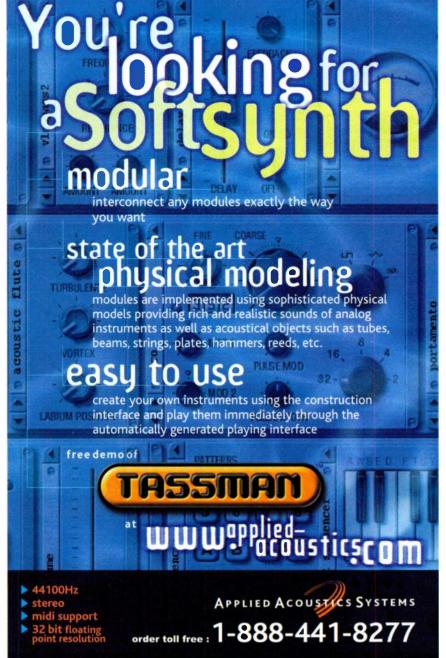
being able to use tap tempo from the D-Two to gain control of sequencers, drum machines, or even an M-One or a second D-Two in a live performance. Still, its customizable rhythm patterns, quantization features, and variety of delay types put the D-Two ahead of the class in its price range.

#### **LIKE MINDS**

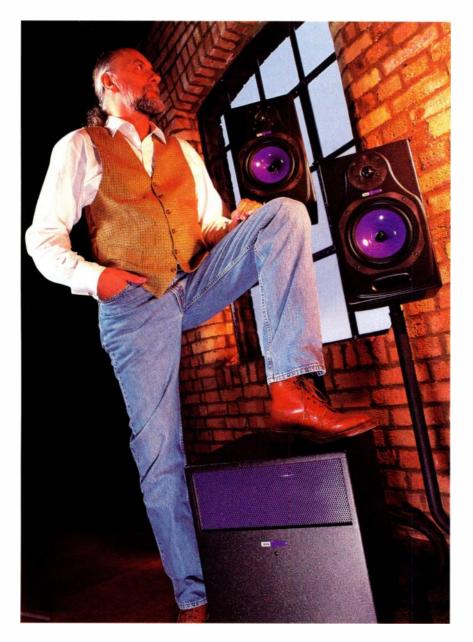
TC Electronic's M-One and D-Two are very powerful yet affordable new multieffects units from one of the most highly respected names in signal processing.
They should appeal not only to budgetconscious personal-studio owners and
live-sound engineers, but also to pros
seeking to expand their processing
racks with premium-sounding yet easily
programmable gear.

Operationally, the two units are very similar: Store and Recall work the same way, both have the same three bypass modes, and all parameters are readily accessible. Moreover, you can control just about every parameter through MIDI, and both models can lock to an incoming MIDI clock and subdivide it to adapt to very slow or fast tempos. I highly recommend the M-One and D-Two.

Music-technology consultant Mike Collins lives in London, where he plays guitar, writes and produces music, teaches music technology, and writes for magazines worldwide about all of this stuff.



# THE RUMORS ARE ALL TRUE





Accurate. Revealing. Detailed. Untiring. HHB Circle Studio Monitors deliver the best-value monitoring solutions in the business.

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Just one session was all it took for the legendary musician Mick Fleetwood to standardize on Circle monitors for his studios in London, Los Angeles and Hawaii. "When I heard that detailed, robust bottom end, I said: 'I love them. Where can I get a pair?'"

Reviewing the Circle 5s, Pro Audio Review concluded "Detail, clarity, off-axis response and imaging are superb... definitely give the HHB Circle 5s a listen" while Audio Media said of the Circle 3s "I fell in love with the active pair as soon as I plugged them in... the Circle 3s sang out loud and proud with a reassuring kick in the gut providing enough bass, while remaining accurate."

So what are you waiting for? Get your ears down to the nearest HHB dealer and check out Circle monitors today.



Above (left to right): HHB Circle 5 midfield monitor (active and passive versions available), HHB Circle 1 powered sub-woofer, HHB Circle 3 nearlield monitor (active and passive versions available).

Left: Mick Fleetwood with HHB Circle 5 active midfield monitors and Circle 1 powered sub-woofer.

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## CAKEWALK

PRO AUDIO 9.0 (WIN)

A best-selling digital audio sequencer moves one step closer to perfection.

#### By Phil Darg

ost desktop music producers throughout the '90s dreamed of a seamlessly integrated MIDI and audio production environment that offered lightning speed and a vast array of features. Today, a number of applications are rapidly turning that dream into a reality.

Cakewalk's *Pro Audio* is one of them. *Pro Audio* has combined MIDI sequencing with multitrack digital audio for several years now, and version 9.0 brings users even closer to living the dream of the virtual studio.

#### NUMBER 9 . . . NUMBER 9 . . .

Pro Audio is an integrated MIDI and multitrack audio recording, editing, and mixing application. Recent versions added features such as real-time MIDI and audio effects; 24-bit, 96 kHz

recording; and the ability to sync video files directly to audio tracks. Cakewalk calls the latest version "Evolution 9," and I can't think of a finer appellation. Version 9.0 continues the company's tradition of improvements in *Pro Audio*'s features and performance, which has made it a leader in MIDI and audio technology.

Seasoned users of *Pro Audio* will find version 9.0's graphical user interface familiar. Very little has changed, with the exception of the Console view (see Fig. 1), which has been given a face-lift. Some of the main menus have been rearranged for efficiency as well. Here, we'll cover the new features and enhancements lying beneath the surface.

#### RIDING THE WAVEPIPE

One of the most important breakthroughs in Pro Audio 9.0 is its new WavePipe technology. WavePipe is a more efficient way for the software to transfer and process audio data. The result is a decrease in audio latency that translates into more-responsive console sliders and an increase in the number of tracks and effects that can be mixed or played back in real time. According to Cakewalk, performance gains vary depending on the sound card you use and the design of its driver. Most users, however, can expect a significant improvement in audio performance.

Value: Chorus All Sound Resident Reside

FIG. 1: Cakewalk *Pro Audio* 9.0 sports an enhanced Console view. Not only are the graphics more attractive, but the application also includes new buttons for auto-numbering tracks, applying effects, and more.

Pro Audio Minimum System Requirements Pentium II/200; 64 MB RAM; Windows 95/98/NT 4.0

How well does the WavePipe technology work? To find out, I ran a number of tests with Pro Audio 9.0 on two different systems: a 233 MHz PC with 64 MB of RAM and Digital Audio Labs' (DAL) CardD Plus, and a 600 MHz PC with 128 MB of RAM and a DAL Card-Deluxe. On the 233 MHz PC running Pro Audio 8.0, I could play back 16 mono CD-quality tracks with no effects-but just barely. After installing Pro Audio 9.0, I was able to play 20 mono tracks of audio with ease on the same PC. This translates into a 25 percent improvement in performance, so even older systems will see substantial gains. In fact, it wasn't until I loaded 22 mono tracks with four real-time effects that I ran into any dropouts at all.

Using version 9.0 on the 600 MHz machine, I managed to play 30 mono 24-bit tracks simultaneously without incident. Even with 30 tracks playing, I was using only 20 percent of my CPU and 60 percent of the PC's disk-transfer ability. Pushing harder, I ramped up the project to 30 mono tracks with eight effects, which seemed to be near my limit. This much processing power is great-for the first time ever, I'm confident that Pro Audio is more than able to deliver the virtual mixing performance that I need. The bottom line is that Pro Audio 9.0 increases flexibility by allowing you to avoid mixing down tracks during a project. This way you can keep your options open until it's time to do the final mix.

Another type of performance enhancement can be gained through the use of AudioX, a new protocol for communications between audio hardware and software. AudioX enables Pro Audio 9.0 to address the additional features that a compliant sound card has to offer. Pro Audio automatically configures a Console for the device after determining the features—such as onboard DSP effects, custom bussing capabilities, or word clock—that it offers. AudioX drivers are currently available for several sound cards, including the Yamaha DSP Factory, the

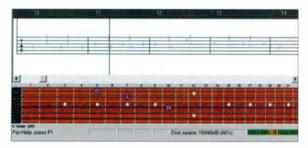


FIG. 2: *Pro Audio* 9.0 offers a number of new features for guitar players, including tablature and fretboard displays.

Digital Audio Labs TDIF 2496 Pro, and the Sonorus StudI/O.

Cakewalk has given version 9.0 additional CPU and disk meters to monitor audio streaming. These small, helpful displays sit unobtrusively at the bottom of the screen and continuously measure CPU power and disk throughput as your tracks play. For example, if you notice that you're close to maxing out your resources-which can cause dropouts—you can bounce tracks or apply your effects destructively to save processing power. If a dropout does occur, a red warning display will come up. Another small display box indicates the amount of free space remaining on your hard drive.

Cakewalk promised smoother audio scrubbing in version 9.0, and it definitely delivers. In earlier versions, scrubbing was often choppy and fraught with dropouts. Not so with *Pro Audio* 9.0. In fact, not only does the audio scrub perfectly, but it does so effortlessly, and you can scrub more than one audio track simultaneously, backward or forward.

#### **CAKE MIX**

Pro Audio 9.0 provides many new options for mixing down and bouncing tracks. In older versions, you had to mix down all of your audio to a pair of stereo tracks and then export those in WAV format to create CD-ready files. Now you can select and export any number of tracks-or any section of a track or tracks-and decide which parameters (effects, volume, and panning, for example) will be included. This feature is a huge improvement and a real time-saver. You can save files as mono or stereo, but even better, you can export your files in MP3, RealAudio, and Windows Media formats, as well as in WAV. (Pro Audio uses Fraunhofer/ Thomson's highly regarded MP3 encoding technology.)

A new Apply Audio Effects button can destructively apply effects to all audio tracks in one fell swoop. Previously, you had to either apply effects individually (which could be quite a chore) or perform a mixdown. Keep in mind that processed tracks replace the original, so it's always a good idea to back

up your audio in case you want to "undo" an applied effect. *Pro Audio* 9.0 also makes more efficient use of the Track view by allowing a single stereo audio clip to fit into a single track. (In previous versions, all tracks were mono.) *Pro Audio* can even mix and match mono and stereo audio on the same track.

#### **VIEW ENHANCEMENTS**

Another improvement is the addition of Record, Solo, and Mute buttons to the Audio view. This is an excellent idea because in past versions, you had to toggle between Track and Audio views to mute and solo tracks. In fact, it's such a great feature that I would like to see it extended to the Staff view as well. Version 9.0 also includes a new global Arm, Solo, and Mute toolbar, which is used to toggle the aforementioned functions on all tracks. For example, if 3 of your 16 tracks are set to Solo, clicking the global Solo button turns off the Solo feature on all 3 tracks at once. However, clicking the button again solos all of the tracks, not just the original 3. The button has its usesfor example, it allows you to arm all

tracks for recording at once. Still, some type of "intelligent memory" feature would make a better addition.

A further enhancement is the ability to assign effects directly in the Track view. (In previous versions, you had to switch to the Console view before performing that function.) This feature "adapts" to the type of effect appropriate for the track (that is, only MIDI or audio effects are shown as choices on tracks of

those types). If you have audio and MIDI on the same track, however, only the audio effects appear. Unfortunately, double-clicking on the Effects field brings up the FX bin, as in the Console view; you must right-click again to access the available effects for that track. After you select an effect, you still don't have access to the effect's editable parameters—that requires yet another double-click. I would prefer the effects list to come up with the first click.

Version 9.0 adds considerable functionality to the Piano Roll view and includes an improved Patch Browser. You can now open, select, and edit multiple tracks of MIDI data simultaneously in the Piano Roll. MIDI data is colorcoded on a track-by-track basis, and any number of tracks can be displayed or hidden. The Patch Browser works from the Track view to help you quickly locate a patch, allowing you to search for it by name.

Guitar players will also find useful enhancements in Pro Audio 9.0. These include a new fretboard display, guitar tablature, and a tuner, as well as StudioWare control panels for the Roland GR-30 and Line 6 Pod. The fretboard and tablature displays can be accessed from the Staff view (see Fig. 2). They work much like the Staff view itself, displaying notes and highlighting them as they play in real time. Guitarists who prefer to read tablature or who would rather follow the notes on the fretboard will get a lot of mileage out of these two features. The onscreen tuner is also useful, although it's limited to DirectSound-compatible audio cards and a 16-bit, 44.1 kHz setting.

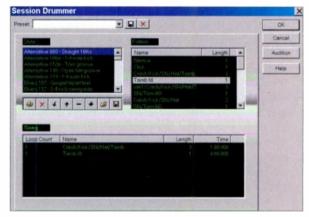


FIG. 3: Session Drummer is a new plug-in that can quickly generate drum tracks in numerous styles. Users can also build their own styles and reuse them across projects.

#### **GET IN STYLE**

One of the most interesting new additions to Pro Audio 9.0 is a "lite" version of Ntonyx's Style Enhancer Micro (SEM) 1.28. SEM is a MIDI plug-in for Cakewalk that uses Performance Modeling technology to generate MIDI controller commands and note patterns. This plug-in can transform data in ways similar to CAL (Cakewalk Application Language) routines, but it has a graphical interface and is much more advanced. In reviewing Pro Audio 9.0, I had the opportunity to look at the full version of this application and was impressed with its ability to increase the liveliness of MIDI tracks in a realistic fashion.

Styles are the key to SEM. A Style is a process that is applied to elements of a track or a range of data and can be as simple as arpeggiating a chord or as complex as adding large amounts of MIDI controller data to help shape a phrase. Each Style includes a description of the type of data with which it is intended to be used (see Fig. A). For example, one Style might be best suited to a singlenote melody at a moderate tempo, while another might be intended to process a sequence of slowly moving chords. (Of course, you can try any combination of Styles and MIDI data you want, but you're not guaranteed to see useful results.) Although I needed some practice to get SEM working effectively, I soon discovered that it does some amazing things.

I tested one of the pianoaccompaniment Styles by applying it to a series of piano chords I had stepped in. The result was an entire track of piano notes—with expression—that fit the chord progression I'd created. I also applied one of the guitar Styles to a guitar part and was impressed with how the plug-in simulated the vibration and decay of a guitar string by adding a slight amount of Modulation Wheel and Expression Controller data to the notes.

Another Style was useful for creating "sliding bass" lines. It added Pitch Wheel data to the part so that notes glided smoothly from one to the next. An entire set of Styles is intended for drum music, including one that can expand the range of dynamics in the part by assigning a random offset above and below the existing notes' Velocity values. I also found that the SEM brass Styles worked particularly well to create simulations of breath action, giving MIDI brass tracks much more expression and life than they had originally.

You can check out many different purchase options and bundles at Cakewalk's Web site (www.cakewalk .com). These include a large library of additional Styles that can be used once you've upgraded to the full Cakewalk plug-in version of SEM. Another version of the software, called simply Style Enhancer (SE) 2.1, functions as a stand-alone product. SE comes with some sequencer functions and offers numerous additional parameters for tweaking Styles and altering data. (You can learn more about SE 2.1 from Ntonyx's Web site at www.ntonyx.com.)

If you've ever said to yourself, "Gosh, my MIDI tracks sound lifeless," you may want to check out Ntonyx's *Style Enhancer Micro* 1.28. It won't put a live band in your basement, but it can sure help your modules sound more realistic.

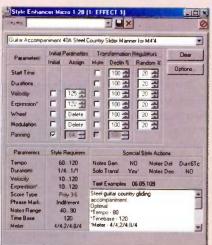


FIG. A: Ntonyx's Style Enhancer Micro uses
Performance Modeling technology to generate
expression and note data for MIDI tracks.
A "lite" version of the program is included with
Pro Audio 9.0 and appears as a MIDI plug-in.

#### THE GOODIES

Other extras in *Pro Audio* 9.0 can add polish to both audio and MIDI tracks. Among these are new MIDI plug-ins (see the sidebar "Get in Style"). The *Session Drummer* plug-in works with *Pro Audio* to generate drum tracks instantly (see Fig. 3). It allows you to choose a style and a specific type of drum clip (such as an eight-bar run or a one-measure fill). The 65 available styles include Jazz. Hip-Hop, Rock, Blues, Dance, Latin, and World, and you can easily add your own styles and patterns. Each style has a number of



#### Cakewalk

Pro Audio 9.0 brings users one step closer to living the dream of the virtual studio.

variations. Session Drummer is a great way to create a drum track quickly. The only real drawback is the time it takes to audition all of the available styles.

Cakewalk has also included its AmpSim Lite package, a stripped-down version of the company's FX2 plug-in. AmpSim Lite models the sound of various amplifiers and offers a choice of "American Lead" or "British Overdrive." Like the included Style Enhancer Micro, it is just a teaser, and many of the settings have been graved out. Even with its limited options, however, AmpSim Lite provides a number of distortion and amp sounds that can be applied to audio. This feature is most useful for dry guitar sounds, but you can also apply it to vocals or even to keyboards. AmpSim Lite can be used destructively or in real time.

#### **JOHANN SUGGESTION BOX**

Although Cakewalk *Pro Audio* 9.0 is a great tool, it could still use some improvements that I would like to see in upcoming versions. First, Cakewalk should include some kind of spectral analysis tool, perhaps even something that could be used in real time. After using such a feature in other audio applications, I realize how crucial it is for zeroing in on mixing problems.

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#### **DANGER, WILL ROBINSON!**

If you intend to purchase or upgrade to Pro Audio 9.0, be sure to check the version number as soon as you get the software installed (select the Help → About Cakewalk option). Users of the original version 9.0 run the risk of losing data if they happen to specify the same directory for storing both their audio data and the PICT files that are used to show the waveform of a file. You can go to the Options → Audio → Advanced tab and change the settings there, if necessary.

If you already have version 9.01 or higher, you won't have a problem. (Further information can be

found at Cakewalk's Web site, where free 9.0 upgrades have been posted.)

Interestingly, the feature that led to this potential hazard also has an upside: Pro Audio now creates WAV files using the standard file extension, which means you can open and play the audio files that you create in Pro Audio outside of the program without having to export them. Before, simply replacing Cakewalk's audio-format file extension WA~ with the traditional WAV extension would render the files unrecognizable by Pro Audio, even when they were converted back to WA~.

(Third-party analysis plug-ins are available for this function.) Also, a MIDI Volume offset that affects all Volume data in selected tracks would be a helpful addition. Currently, changing the volume on the Track view won't work if there are variations of volume inserted into the track itself. You can, however, use a CAL (Cakewalk Application Language) script or the Interpolate command to achieve this effect.

There is one area in which Pro Audio 9.0 seems to have taken a step backward: the Remove Silence function. In previous versions, audio events retained their integrity when this pro-

cess was applied; that is, they remained in place. In the new version, the Remove Silence command actually splits and deletes sections of the audio event, making it more difficult to edit or move events in the Audio view than it was in the past.

Finally, I would like to see some of Pro Audio's bundled audio effects upgraded. Although the Parametric EQ has been improved, most effects remain unchanged from recent versions. I would suggest adding some new effects, such as a phaser, a wah wah, an intelligent harmonizer (for both MIDI and audio tracks), and a 30-band graphic stereo equalizer.

Overall, Pro Audio 9.0 is a significant upgrade that brings this application one step closer to the dream of seamless desktop audio production. Users of version 8.0 and earlier should upgrade immediately. (Trust me, don't even think twice.) For new users, Pro Audio 9.0 is a complete and easyto-use MIDI and audio application that can take a project from conception to CD-ready WAV file. It has more than enough features to keep both amateurs and professionals busy for a long time. Take it from me: before you've exhausted the possibilities of Pro Audio 9.0, Cakewalk will have released version 10.

Phil Darg is an independent composer and producer. His latest work is the instrumental jazz CD Powder Blue Tux and Empty Arms.

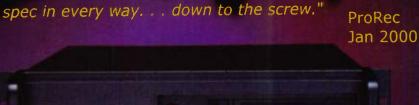


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# LEXICON

**MPX 500** 

Filling the gap between low and high in the formidable MPX series.

#### By Barry Cleveland

t \$599, the MPX 500 multi-effects processor costs twice as much as Lexicon's entry-level MPX 100. That may seem like quite a price vault, but keep in mind that this unit costs only half as much as the flagship MPX 1, making it attractive to those who yearn for more features and programmability yet hesitate to drop more than a grand on an effects processor. Despite its moderate price, the MPX 500 is powered by Lexicon's proprietary Lexichip (the same chip used in the costlier MPX- and PCM-series effects processors) and boasts 24-bit digital converters and 24-bit internal processing. Additionally, S/PDIF digital inputs and outputs let you connect the unit directly to digital mixers, hard disk recorders, and other digital devices, bypassing the converters altogether.

Rather than taking the every-effectimaginable-and-then-some approach common to many multi-effects units, the MPX 500 offers a maximum of two simultaneous effects that are configurable in a variety of useful ways. It also provides fairly extensive programediting capabilities, including four dedicated Edit knobs for near-instantaneous tweaking. The emphasis is clearly on usable presets, with 240 preset programs that need little or no tweaking and cover a wide variety of effects—an important feature when there are only 30 user-program memory locations.

#### **OUTSIDE THE BOX**

The MPX 500's user interface is a marvel of simplicity (see Fig. 1), consisting

of six knobs, six buttons, and, in the center, a 150-by-32-pixel backlit LCD screen. To the left of the screen (left to right) are the Input Trim knob, buttons for accessing Edit Pages and entering System mode, and the four Edit knobs. To the right of the screen are the Load and Bypass buttons, the Program knob, and buttons for the Learn function (labeled Store and Tap/Cancel). All of the buttons have embedded LEDs that light up when engaged; especially snazzy is the LED on the Tap/Cancel button, which flashes in time to the current tempo. A Power on/off switch on the far right completes the front-panel controls.

On its rear panel, the MPX 500 sports a range of input and output options (see Fig. 2). Analog I/O includes stereo pairs of balanced 1/4-inch TRS and XLR connectors; the S/PDIF digital I/O uses coaxial connectors. A 1/4-inch TRS input allows you to hook up a dual footswitch to control the bypass and tap-tempo functions, as well as MIDI In and software-selectable Out/Thru connectors. The unit has a built-in power supply with a detachable IEC power cable.

#### **DANDY INTERFACE**

Impressively, the unit's LCD screen presents a wealth of information in a clear and relatively uncluttered manner. Look for and you'll easily find the name, number, and bank location of the currently loaded program and its routing configuration (displayed as a small graphic). The current tempo in bpm, and whether it's global or program specific, shows up in bold relief, as do the names of up to four editable parameters. On the display you'll also notice a stepped bar-graph stereo input meter that indicates whether the input source is analog mono left or right, analog stereo, or digital (always stereo); if digital, it will tell you whether the sampling rate is 44.1 or 48 kHz. Finally, if you're not using the program AutoLoad feature (more on that later), the display communicates the number of the cued program. As a

bonus, the LCD screen contrast is adjustable via a small pot on the unit's rear panel.

The MPX 500's programs are arranged in banks and selected by using the dedicated Program knob. Turn the knob and you'll march through the programs sequentially; press it and you'll jump to the next bank. Simultaneously pressing and turning the knob—which can be a little tricky until you get the hang of it—allows you to glide easily from one program to another that is hundreds of numbers away with a quick twist of the wrist.

When a program other than the currently loaded one is selected, the LED in the Load button lights up, and for four seconds the name of the selected program replaces the name of the loaded program in the display. You can then load the selected program immediately or anytime thereafter by pushing the Load button. If after four seconds you choose not to load the selected program, then its number (next to the word cued) will pop up under the name of the currently running program. An AutoLoad feature causes programs to load automatically 0.75 second after the Program knob stops turning.

Each program on the MPX 500 can have as many as 16 editable parameters, arranged four at a time on Edit pages that correspond directly to the four Edit knobs. Though the names of the first four editable parameters are visible within the basic program display, their values are not. As soon as one of the Edit knobs is adjusted, Edit page 1 appears, showing the names and values of the first four parameters. The name and value of the selected parameter are highlighted, making it easy to know exactly what you are doing. The pages can also be accessed by pushing the Edit Pages button, which steps you through each page sequentially. Once any parameter value has been changed, the LED in the Edit Pages button lights up as a reminder.

Edit knob 1 is referred to as the Adjust



FIG. 1: The MPX 500 is a well-designed and affordable effects processor with great-sounding presets and an intuitive interface.





FIG. 2: The rear panel provides balanced analog (available on two types of connectors) and digital I/O.

knob on Edit page 1. In many cases, rather than accessing a single parameter, this knob adjusts a combination of critical parameters simultaneously. For example, in many chamber and room programs this parameter is called Liveness, and adjusting it changes EQ, decay, and early-reflection parameters. The name of the Adjust parameter is always shown in parentheses, making it easy to locate; when you adjust it, a more detailed description of its function appears briefly along the bottom line of the display.

#### **ALL SYSTEMS GO**

Parameters that affect the MPX 500 globally (including MIDI settings, MIDI dumps, and reinitialization commands) are accessed by pushing the System button. Examples of System parameters include Output Level (0 dB/unity gain to -24 dB), Input Source, Clock Source (internal 44.1 or 48 kHz, or external S/PDIF), Mix mode (global or program), Bypass mode (dry/full mute/input mute), Tempo mode (global or program), and Program Load mode (which determines whether the dry signal continues or there is silence when you switch between programs).

Two of the more interesting System options are found under the Digital Output and Operating modes. By setting the Digital Output to Dry, you can use the A/D converter independently of the effects processor. The Operating modes include Normal; Demo, which continually loads one program after another; and Locked, which makes only the programs in the User Bank accessible, disables the Edit and Learn functions, and selects the AutoLoad feature. The Locked mode could be handy when you want to access only the user programs without navigating through the other functions. However, I could not get it to work properly at first. Lexicon looked into the problem and discovered an oversight in the user guide: you have to actually load a user program before engaging the Locked mode-a critical step that somehow got left out of

the documentation. After learning the correct procedure, I was able to get the Locked mode to work just fine.

The MPX 500 operates on any of the first 16 MIDI channels and recognizes MIDI Program Change (100 to 127 or 101 to 128), Bank Select, Pitch Bend, Aftertouch, and continuous controller (1 to 31 or 33 to 119) messages, allowing it to be operated remotely from a MIDI sequencer or other MIDI device. You can easily assign any of the MPX 500's editable parameters to a continuous controller (and a controller range can be fixed) by using the Learn mode. The MPX 500 also recognizes MIDI Clock messages and applies a tempo of 40 to 400 bpm to any program featuring tap tempo. Tempocontrolled delay and modulation rates lock to tap or MIDI Clock, and tap tempos can be controlled by audio input, the front-panel Tap/Cancel button, a footswitch, an external MIDI controller, or a MIDI Program Change message.

#### **BANK ON IT**

As mentioned previously, the MPX 500 provides 240 preset programs, as well as memory locations for 30 user programs. The user programs cannot be "built from scratch" but instead are simply modified versions of presets.

Changes made to presets cannot be saved to preset locations (that's why they're called presets), so even if you change only a single parameter ever so slightly, you'll have to save your new "user program" to one of the 30 memory locations. This could be a serious limitation if you need to have access to more than 30 edited programs at once. User programs can be backed up to a librarian program and recalled via MIDI, but that may not be a satisfactory solution for everyone. You can easily name the user programs using two of the unit's Edit knobs.

The MPX 500's programs are organized by type into 25 banks. For example, plate reverbs are located in bank 0, gated reverbs in bank 1, hall reverbs in bank 2, and so on. The first 104 preset programs use single-effect algorithms, whereas most of the remaining presets are dual-effect programs. The dual programs combine two effects algorithms in one of four different routing configurations (see Fig. 3): Dual Stereo or Parallel; Cascade or Series; Mono Split, which routes one side of a stereo input into each effect and combines their stereo outputs; and Dual Mono, which is the same as Mono Split except that each effect outputs discrete mono.

Banks 0 to 6 contain plate, gated, hall,

#### MPX 500 Specifications

Preset Programs	240
User Locations	30
Analog I/O	(2) balanced ¼" TRS; (2) balanced XLR
Digital I/O	24-bit S/PDIF (coaxial RCA)
Other Connections	MIDI In, Out/Thru; ¼" TRS footswitch jack
A/D Converters	24-bit sigma-delta (20 Hz-20 kHz, ±1 dB)
D/A Converters	24-bit delta-sigma (20 Hz-20 kHz, ±1 dB)
A/D Dynamic Range	105 dB
D/A Dynamic Range	101 dB
A/D and D/A Crosstalk	−96 dB @ 1 kHz
Internal Processing	24-bit
Sampling Rates	44.1, 48 kHz (selectable)
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1800 Green Hills Rd., Scotts Valley, CA 95067 Tel: 831.438.1921 Fax: 831.438.8812 www.emu.com room, chamber, and ambient reverbs. Digital reverb has been one of Lexicon's specialties for decades, and these complex yet transparent programs are fine examples of the Lexicon sound. Some of the reverb programs are nearly identical to those found in the MPX 1, and a few even have the same name.

The plate reverbs range from thin and metallic-sounding to big, full, and rich. (Oddly, Tape Slap, at both 7.5 and 15 inches per second, is included in the plate category.) The gated reverbs include a straight gate, several tap-predelay gates, and two inverse gates. You can choose to play in a small, midsize, large, or concert-size hall; in a small or large church; or in a special hall for jazz, dance, or synth music. The unusual and interesting Gothic Hall program has 418 milliseconds of predelay and 2.4 seconds of decay; the Adjust knob controls predelay, which can be extended out to more than a second, resulting in bizarre echo effects.

The room programs emulate acoustic spaces ranging from Bedroom and an excellent Tiled Room to the immense Studio A and the dense Fat Space and Chunky Space programs. Chamber reverbs include a nice Basement and two good vocal chambers, along with a Brick Wall that's not particularly convincing and a great PCM60 Large Chamber. The ambient programs are versatile as well as relatively authentic, with Guitar Amb and Marble Foyer being my two favorites.

Bank 7 contains five tremolo programs, which are differentiated by the use of various modulation waveforms: sine, rectified sine, square, triangle, and sawtooth. You can control modulation speed using the Tap function, and the Adjust knob changes the phase relationship between the two sides of this stereo effect. These are some of the best-sounding tremolo effects I've heard anywhere, apart from old Fender

tube amps. The rotating speaker programs in bank 8 were much less impressive but still proved to be passable versions of this effect, which is extremely difficult to emulate.

The five chorus and five flanger programs in banks 9 and 10, respectively, are quite good. The six-voice choruses are based on those found in the PCM80, and two of them add a little slap delay before the modulation. Three of the flangers are configured in a 180-degree phase relationship, with the Adjust knob controlling resonance. Bank 11

The MPX 500's user interface is a marvel of simplicity.

contains five detune programs, which are voiced similarly to the choruses but are more dramatic and exaggerated. The five stereo polyphonic-pitch programs in bank 12 provide pitch-shifting of two octaves down and one octave up, with adjustments in fine increments. The pitch algorithms sound really good (especially when combined with other effects in the dual-program banks), and I detected only negligible tracking errors.

Bank 13 contains 15 delay-and-echo programs, all of which sound exceptionally clean and crisp. Tap tempo sets the delay time, and the Adjust knob sets the amount of feedback on the first eight programs, whereas on the other seven the Adjust knob sets the delay time. Mono delays can be up to 5.5 seconds, and stereo delays can be half that length. The delay programs feature a very clever Master Delay parameter that scales all non-

tap delay times within a program by a specific percentage.

Bank 14, special effects, contains 15 intriguing programs employing a variety of configurations. Fans of the Infinite reverb program found in the discontinued LXP-1 will be pleased to find it here. It's basically a chamber with an infinite yet distant decaythink Paul Horn in the Taj Mahal. Infinite Delay links stereo delay times to tempo and loops back on itself with 100 percent feedback. Abyss and Dreamscape both run pitch into reverb; the former feeds a subtle detuning algorithm into a sizzling reverb with a long decay, and the latter generates cascading harmonies into a large reverb with medium decay. Jet Flange lives up to its name in every way, particularly with the resonance cranked up.

The remaining banks contain the dual programs, which combine reverb and delay with one of the other individual effects algorithms. In each of banks 15 to 21, the first six programs are configured as Dual Stereo, and the last four as Cascade. Banks 22 and 23 are configured as Mono Split, and bank 24 is configured as Dual Mono.

#### **APPLY AS NEEDED**

I had the opportunity to use the MPX 500 while tracking and mixing in my personal studio, and I found it to be quite useful in both applications. I achieved excellent results when connecting it to my Yamaha 03D digital mixer using the S/PDIF digital output, and it performed nearly as well using either set of analog outputs. In all cases the sound was sharp and clear, with practically no noise or digital artifacts.

I also used the MPX 500 in my guitar rig while performing live. Normally, I run a stereo guitar preamp into an MPX 1 and then connect the MPX 1 to a stereo power amp. When I hooked up the MPX 1 to the MPX 500 via the

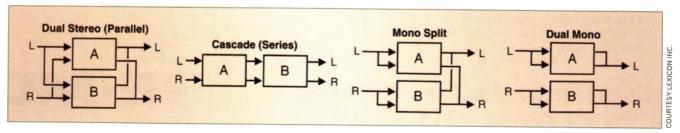


FIG. 3: This diagram illustrates the four ways that dual programs can be configured on the MPX 500.

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CONS: Has only 30 user-program locations.

S/PDIF connectors, the two units worked together like a sort of super-processor, allowing me to create ultra-complex effects combinations. For example, by using one of the MPX 500's pitch algorithms to process a harmonizer program in the MPX 1, I created massive chords with decidedly otherworldly overtones.

Being able to quickly access and adjust the main parameters of a multieffects processor by simply turning a
knob or two is a great advantage when
working against the clock in a studio—
but being able to do it while performing live is nothing short of fabulous. I
work with an improvisational ensemble, so my ability to change sounds
spontaneously at less than a moment's
notice is critical. The MPX 500 gives
me that flexibility.

#### **SHORT AND SWEET OF IT**

The MPX 500 delivers the latest digital technology, extensive I/O, and intelligently conceived and great-sounding programs at a reasonable price. It is ideal for users who need good working presets that can be tweaked to fit quickly and easily, but who don't necessarily require the capacity to store lots of user programs onboard.

Barry Cleveland is the editor of Mix Master Directory and The Recording Industry Sourcebook, and author of the "Pedal Board" column for Onstage. He also plays guitar in the improvisational quintet Cloud Chamber (www.mphase.com/cloud.htm) and operates a personal studio.

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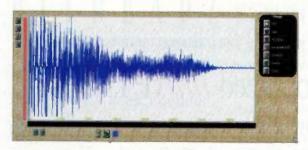


FIG. 2: You can edit and process individual drum sounds in Voodoo's waveform editor.

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Voodoo has four floating windows: Transport, Meters, Tool Bar, and Keyboard. The Transport window has everything you'd expect-Play, Record, Stop, Rewind to Last Pattern, Forward to Next Pattern, and Loop Playback. Above the controls, an easy-to-read display shows the elapsed time in beats and measures. The Meters window shows the left and right master levels and effect 1 and 2 outputs. The meters aren't particularly informative (the effects meters are mono even though their returns are stereo), but they do let you see the signal. The Tool Bar window provides access to a dozen

commands, such as Cut, Copy, Paste, All Notes Off, and Record to Disk. I liked it better than searching through the main menus but found the icons confusing; popup labels would be a big help. The fourth window, Keyboard, is handy for checking keymaps, locating samples with the mouse, and sending Note

On messages directly from Voodoo.

Using a mouse to adjust software sliders and knobs is a pain, but Voodoo's are particularly well designed and more mouse-friendly than most. All controls can be adjusted in large or small increments depending on where you click: clicking and holding a knob on the top or bottom lets you step through values in small increments; clicking and holding on the left or right sides allows you to zoom through values in large increments. Sliders work in a similar fashion. And every control has a numerical readout, which is a big help when making precise adjustments.

Double-clicking on the readout highlights it and allows direct text entry; this is a handy option for inputting specific numbers.

Voodoo's main window takes a while to get used to. The program packs a lot of information onto the screen at once, which is nice in some ways, but it can feel cluttered. I'd prefer to have discrete windows that I could organize to best fit my creative flow; if I'm done adjusting the pad parameters, for instance, I don't want those controls taking up screen space until they're needed again. Voodoo does allow you to partially redesign its windows: you can change the background colors, replace the knobs, and alter the sliders (see Fig. 3). It's all very nice, but superficial-you can't physically reposition control elements within the windows themselves; you can only change their appearance.

#### SAMPLE THIS

BitHeadz packages Voodoo with a number of excellent kits, each of which sounds solid and may be all you need for years of blissful beat making. The kits include such standards as R&B,



electronic, rock, orchestral, and Latin music. All are sampled with 16-bit resolution, and most employ a 44.1 kHz sampling rate. The Vin Time Traveler 1 kit is especially nice, with its phat kick and old-school flavor. The Justin Time set has beautiful deep-house percussion and effects. Latin Percussion has a sweet Velocity-zoned a-go-go. The Construction Kits folder even includes a batch of cool kits based on loops, such as straight-ahead drum loops, bass lines, fills, sound effects, and some smooth ride-cymbal patterns. (I'm always looking for ride loops because single ride hits played in eighth-note patterns never sound real.)

BitHeadz's samples impressed me enough to go looking for them on my hard drive to use as audio files for a remix I was working on in Pro Tools. I searched diligently, to no avail. It turns out that the samples are embedded in *Voodoo*'s files. However, they can be extracted by highlighting the waveform in *Voodoo* and then using the Export Sample command.

Voodoo also comes with a bunch of Standard MIDI Files—including reggae, samba, waltz, dance, and jazz—that function as demos for the drum kits and as patterns for song construction. I wasn't particularly impressed with any of the files, but I rarely use preset patterns in my productions anyway. I'm sure they'd be fine for writing a song or working out a demo arrangement. Standard MIDI Files can be dragged and dropped directly into Voodoo's Pattern list from the desktop, which makes auditioning files a snap.

#### **RHYTHM PATTERNS**

Recording your first pattern is a simple matter of selecting a tempo and popping *Voodoo* into Record mode. After the first pattern, you use the Add Pattern command to create more pat-

#### Voodoo

Minimum System Requirements
Mac: 604e/200; 64 MB RAM; 250 MB free
hard disk space (for full install); Mac
0S 7.6.1 (Mac 0S 8 recommended)
PC: Pentium II/200; 64 MB RAM; 250 MB
free hard disk space (for full install);
Windows 95/98; DirectSound-supported
audio card (multiwave sound card
recommended)

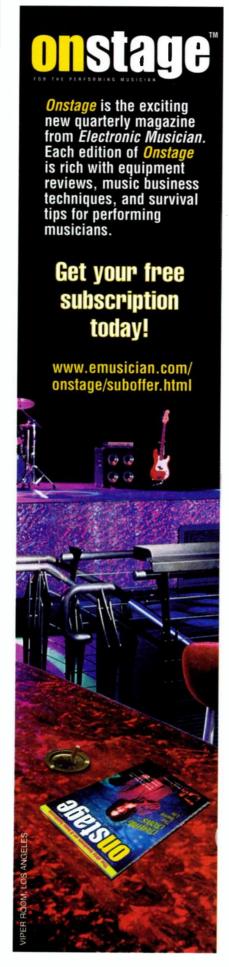
terns. An onboard metronome provides a two-bar count-off. Meters and pattern lengths are set during a pattern's initial recording, but they can be changed at any time.

You can assemble several patterns into a playlist in the Pattern list area to create a song. The playlist order is easily rearranged by clicking and dragging the patterns within the display. Each pattern can have its own number of repeats, and the repeats are automatically played as part of the playlist. For example, if a pattern appears only

once in the playlist but is set to repeat eight times, the playlist will loop that pattern eight times before moving on to the next pattern. Song construction with *Voodoo* is a piece of cake.

When a pattern is selected, its notes appear in the Event Editing window at the bottom of the main window, which can be switched between an event list and a graphic editor. The window is rather small, however, and cannot be resized except by enlarging the entire main window. Default tools include a hand, for selecting and moving notes,





and a pencil, for adding notes. When the cursor is over a note, it becomes the hand; when it's not over a note, it's the pencil. Holding down the computer's Control key turns the cursor into a crosshair for grabbing regions, and holding down the Option key produces an eraser tool. Entering notes and values in the event list is a bit awkward, though. You can't, for example, use your keyboard controller to change a note; you must enter the note from your QWERTY keyboard. Neither can you zoom in or out on the graphic editor.

Commands for selecting specific events, transposing notes, changing Velocity values, quantizing, and using the Go To function are accessible from a column of buttons to the right of the Event Editing window. With this toolbar, along with the graphic editor and the event list, you can edit just about everything. Only one Swing setting is provided, however, so don't count on this sequencer to make your grooves feel better. You'll have to do that the old-fashioned way: by playing the rhythms the way you want them to sound. On the other hand, you can quantize notes while recording, during playback (the recorded positions aren't altered), by pad, by range, or individually. One level of undo is available, and the sequencer's resolution is 960 ppqn.

The onboard sequencer is quite complete for a program of this type, more so than on any drum machine. However, it's certainly no competition for a full-featured dedicated sequencer, mainly because its single-track design and small editing display make it awkward to use. I was also disappointed to

learn that the sequencer doesn't record continuous controller data. Yet with Voodoo's ability to have up to 16 of its parameters—including filter cutoff, delay times, pitch, and LFO speed—piloted by Control Change messages, the program offers a lot of soundwarping power. Unfortunately, because the sequencer doesn't record these messages, you can't automate the parameters from within Voodoo. (Of course, you can always use an external sequencer; more on that later.)

#### **IN EFFECT**

Voodoo's two real-time effects can be used simultaneously. Each pad has an associated send for each effect, providing you with complete control of the wet/dry ratio for every instrument. Master sends, a master return, and a master pan control are available for both effects.

Either effect can be one of three types—delay, reflection, or reverb—and you can tweak three parameters per effect type. Delay has two delay times and feedback. Reflection offers predelay, brightness, and length (the number of reflections to decay). Reverb has predelay, brightness, and decay parameters. Effects presets are not provided, but the effects changes that you make are saved with the kit, and the effects are all in stereo.

A different batch of effects—reverse, gain change, EQ (parametric and shelf), flange, and delay—is provided for destructive editing. The effects are monophonic, but processing a stereo file does not change its format to mono—so in the end, the processed file

sounds stereo because its original imaging is preserved. Reverse is nice for creating weird, sweeping, backward effects. Gain change lets you alter a sample's amplitude from 0 to 200 percent. The parametric and shelf EQs aren't the cleanest I've heard, but they get the job done. The flanging effect is pleasantly raw and dirty. The delay is similar to the real-time delay but in mono.

Voodoo's effects are not particularly impressive; they're sort of lo-fi,



FIG. 3: With a little time and energy, you can use Adobe's *Photoshop* and Apple's *ResEdit* to redesign *Voodoo*'s main window. Here's my own techno-flavored creation.



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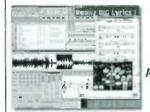
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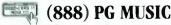
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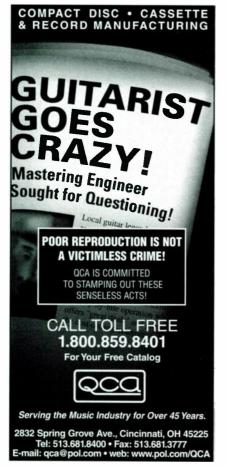
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though they work fine for most soundmangling experiments in hip-hop, electronica, trance, and similar genres. If you need big, lush reverbs; clean, pure delays; and warm, precise EQ, stick with third-party plug-ins—or use a good external signal processor.

#### **HAVE IT YOUR WAY**

One of the best things about *Voodoo* is its well-rounded set of sound-design features. Any AIFF, Sound Designer II, *Unity DS-1*, or audio-CD file can be imported and assigned to a pad, allowing total sound-set creation from the ground up. All but the audio-CD format (which must be imported) can be dragged and dropped from the desktop straight into *Voodoo*'s waveform editor. From there, the sample is automatically assigned to the currently selected pad.

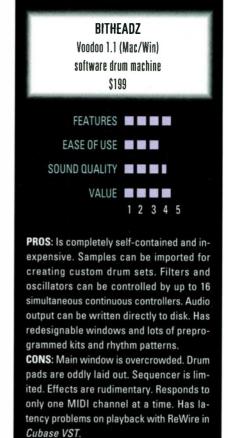
Each pad has up to four Velocity layers, allowing it to hold up to four samples. Layers are easily selected for editing with a vertical bar on the right side of the waveform editor. The layers are in different colors, making their zones easy to identify. Two mute groups are also available for assigning instruments that use the same voice—as, for instance, when you want open and closed hi-hats on different pads to share a single voice. Mono and stereo samples are supported.

The waveform editor works fine for quick cutting and pasting and for simple looping (for example, forward loops with no crossfading or bidirectional tricks), but more-complex operations like normalizing, phase inverting, and plug-in processing are best done with a dedicated editing program. Voodoo's editor is difficult to work with because its output is routed only to Sound Manager. Even though the pads' outputs can be assigned to your sound card, you can audition the waveform on the editor only through the Mac's minijack output. (Voodoo's onboard metronome has the same problem.) According to BitHeadz, the next *Unity* DS-1 engine update (version 2.0) should be out by the time you read this, and it should fix the problem.

Each pad has four sound-shaping parameters: a tuning oscillator, an amplitude envelope, a resonant filter, and an LFO. The tuning oscillator is used for keymapping and tuning samples; the pitch can be controlled by a continuous controller of your choice but

not by the Pitch Bend wheel. The amplitude envelope offers full-fledged ADSR envelope shaping, and it even includes a sustain decay stage. I love the resonant filter for its gritty, pronounced character. It's comprehensive with lowpass, bandpass, and highpass options, and you can set a variety of parameters, including distortion, envelope shape, resonance, cutoff, and Velocity control. The LFO, which is great for creating demented ambient percussion sounds, can be tied to the pitch, filter, or amplitude parameters using sine, triangle, square, or sawtooth waveforms.

An oddball feature called Drum Fill is hard to figure out. I expected it to create a fill automatically when I hit a pad; in other words, I figured that hitting, say, the snare drum pad would trigger a snare fill. That was not the case. Instead, Drum Fill is meant to work specifically with sampled drum fills, not with individual hits. Hitting the pad with the sampled fill should trigger the sample in time with your beat (for example, on the upbeat of 3). However, *Voodoo*'s documentation doesn't reveal where these sampled drum fills are located, and I couldn't find them. BitHeadz



has suggested that it may drop this feature from *Voodoo* altogether.

#### **BLACK MAGIC**

To automate Voodoo, you must use an external sequencer to record the continuous controller moves. In theory, this is a great idea; in practice, it's a bit of a pain, because the program responds to only one MIDI channel at a time. I was able to lock up Voodoo with Steinberg's Cubase VST digital audio sequencer and record Mod Wheel data for controlling resonance. Everything seemed to work fine. But on playback, I discovered that every drum with the resonance filter turned on was getting the Mod Wheel message. Needless to say, the result was not pretty. I wish there were some way to assign each of Voodoo's 14 pads to its own MIDI channel. A multichannel mode instead of a single-channel poly mode should solve the problem. In the meantime, controllers have to be used judiciously.

I did run into a bit of a latency problem when using Voodoo with ReWire in Cubase VST. The trouble became apparent only during playback of alreadyrecorded MIDI events. I didn't notice any latency when I just played Voodoo by itself or when I routed it through the VST Channel Mixer. But when I compared a cowbell playing downbeats to Cubase VST's audio click (also playing downbeats), I noticed a significant delay. Delaying Cubase VST's click until it was locked with Voodoo's cowbell took an adjustment of about 800 samples a substantial amount. I spoke with a BitHeadz representative about this latency, and he attributed it to an audioplayback buffer problem. Apparently, Voodoo's audio is delayed from the time it gets the MIDI note to the time that Cubase VST and ReWire let the audio pass through the VST Channel Mixer. The rep went on to say that ReWire's developer, Propellerheads, was currently working on tightening up the Voodoo-ReWire-Cubase VST connection. He was quick to point out, however, that the problem originates with the way that ReWire and Cubase VST handle the audio and is not inherent to Voodoo. Furthermore, the degree of latency is to some extent CPU dependent and may also be affected by various parameter settings, so your results may vary from mine.

When you do run out of DSP power—which may happen if you use *Voodoo* 

with Cubase VST and lots of plug-ins—Voodoo can export its master output (complete with effects) directly to your hard drive as an AIFF file. The program can do this on its own—without Cubase VST—or through the VST Channel Mixer. I love this feature because it lets you free up processing power for other tasks. Just drop the AIFF file back into your project and keep working.

#### **VOODOO YOU DO**

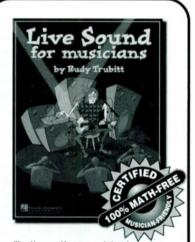
This program is deep. It's impossible to cover all its features in a single review. Suffice it to say that it offers good bang for the buck. With onboard effects, lots of solid drum sets, loads of Standard MIDI Files, powerful envelope controls, and the ability to import samples for creating your own kits, this is some mean voodoo. I could argue that Voodoo's user interface is a bit annoying in areas, but that's a minor point considering the power this software provides. Besides, I've become spoiled from using programs that cost significantly more than Voodoo. Most people will find the interface quite adequate.

My main gripe—the latency I encountered on playback with Cubase VST—may not be Voodoo's fault, but it's something that needs to be remedied quickly. The potential for an excellent partnership between these two programs is exciting. Imagine using Voodoo with Cubase VST's groove templates and VST effects without worrying about timing problems, and a whole world of creative possibilities opens up. Voodoo also works with Digital Performer and Logic Audio, but I wasn't able to experiment with either of these applications.

If you're looking for an inexpensive stand-alone virtual drum machine, you're in for a treat with *Voodoo*. If your goal is to seamlessly integrate *Voodoo* with a high-end sequencer, it may not be your dream program, depending on your other software. Nonetheless, it's hard to go wrong at this price, even if you do have to nudge MIDI events, delay audio buffers, and export *Voodoo*'s audio output to disk to get a tight track. After all, it's not as if we don't do crazy stuff like that already.

Erik Hawkins is a musician/producer working in Los Angeles County and the San Francisco Bay Area. Visit his Web site at www.erikhawkins.com for more equipment chitchat and tips on what's hot for the personal studio.





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# WAVES

NATIVE POWER PACK II (MAC/WIN)

Potent plug-ins that squeeze, filter, and enhance your tracks.

#### By Rob Shirak

f you are even remotely serious about working with digital audio on your desktop computer, you'll find Waves' Native Power Pack II plugin bundle to be an essential musical tool. It doesn't matter what plug-in format your software supports—TDM, MAS, VST, or DirectX—Waves has designed some of the finest mini apps available at any price.

The original Native Power Pack—another indispensable bundle—approaches audio manipulation in a precise, digital manner, whereas Native Power Pack II has more of an analog feel. The newer bundle consists of four plug-ins: DeEsser, MaxxBass, Renaissance Equalizer, and Renaissance Compressor (the TDM Power Pack II bundle also includes PS22 StereoMaker). Not only do these plug-ins complement the original Native Power Pack collection, but they also can compete seriously with some of the finest analog hardware.

As with all Waves plug-ins, a single in-line dongle serves as copy protec-

tion for the software. This lets you install the plug-ins on several computers; just remember to take the dongle with you from machine to machine.

#### FROM HIGH TO LOW

The MaxxBass plug-in is designed to add more perceived bass energy to individual tracks or mixes. Not to be confused with a subharmonic synthesizer (which artificially creates frequencies an octave or two below the original signal), MaxxBass creates harmonics above the original bass signal. This creates the illusion of deeper, fuller bass on smaller playback systems, such as home stereos and TV sets. Used judiciously, it can ensure that full-frequency mixes designed for large monitoring systems (like dance music) translate well to close-field monitors, too.

Small speakers, including many close-field monitors, are not physically capable of reproducing the fundamental frequencies of most bass notes in a full-range mix. Nevertheless, the brain perceives these low frequencies due to a psychoacoustic phenomenon that re-creates those missing fundamental notes from their upper harmonics. Precise control of these upper harmonics is the key to MaxxBass's ability to give a track a stronger and more vivid bass presence.

A graphic display shows the original bass and the created harmonics, which is useful for determining the settings of the various parameters (see Fig. 1). The Frequency slider determines the crossover point; frequencies below the

cutoff will have harmonics created for them, while all frequencies above the cutoff are passed without modification. Volume sliders control the level of the input, the original bass signal below the cutoff, and the newly created harmonics. You can completely remove the original bass signal, leaving only the harmonics MaxxBass generates. (This is the preferred method

# Native Power Pack II has more of an analog feel.

for specific playback installations incapable of reproducing the original bass fundamentals, because it reduces the burden on the amps and speakers.) A monitor section lets you listen to the full audio output, the *MaxxBass* harmonics only, or the original bass frequencies only. This is extremely helpful in determining the plug-in's effect on the audio.

A three-position HighPass button applies filtering to the generated harmonics; the main graphic window also displays changes to this filter. Position 1 is a fixed 24 dB/octave rolloff at 16 Hz (mainly for DC removal), while positions 2 and 3 are 12 dB/octave and 24 dB/octave rolloffs, respectively; the cutoff is determined by the Frequency setting.

The Decay setting determines the output profile of the generated harmonics. Higher values create more upper harmonics. Setting too high a value can cause muddiness in full mixes; setting too low a value can render the effect imperceptible on small speakers. You can apply a rudimentary compressor to the created harmonics to smooth the signal a bit, and a Response setting controls the attack and release of the created harmonics.

You can use MaxxBass for final-mix processing or individual instrument tracking. If you use it on individual instruments, be sure to monitor in context with the rest of the track to ensure proper harmonic levels. I used MaxxBass mostly on final mixes to give them the same low-frequency power and depth on close-field monitors that I was hearing and feeling on the larger speakers.



FIG. 1: *MaxxBass* adds more perceived bass energy to a track by generating harmonic material above the fundamental bass frequency.



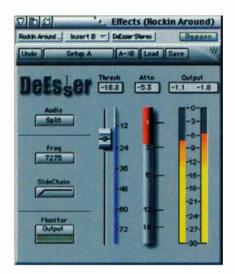


FIG. 2: DeEsser's streamlined interface greatly simplifies the process of removing sibilance.

When set properly, MaxxBass won't greatly affect the bass perception on large studio monitors, but it will make the close fields and mono cubes happen in a big way. Of course, if you are mixing for a specific playback medium, MaxxBass can be a lifesaver for your mixes.

Careful listening, comparisons of the original bass and harmonics, and experimentation are necessary to become proficient with *MaxxBass*, but it is definitely worth the trouble. The first time you hear the bass on one of your mixes slamming on a little Auratone, you'll be hooked. Caution is advised, though—don't overdo it.

#### SUFFERIN' SUCCOTASH

Whereas MaxxBass adds perceived bass energy, DeEsser concentrates on removing nasty high frequencies. DeEsser provides plenty of control with a clear, uncluttered interface (see Fig. 2). The Threshold parameter determines the level at which the de-essing affects the input, with a numeric and graphic display of attenuation and output level. DeEsser operates in a fashion similar to that of complex de-essing chains. The input signal is split into an audio path and a sidechain. The sidechain employs a filter, an energy detector, and a compressor. The sidechain filter has two modes of operation: Highpass and Bandpass. In Highpass mode, a broad range of high frequencies is selected for de-essing, while Bandpass mode enables you to select a more specific frequency range. The Frequency setting, which determines the center point of de-essing, can be set anywhere between 2 kHz and 16 kHz. It affects only the sidechain response, not the audio.

After you've set the Frequency, SideChain, and Threshold controls, the attenuation is applied to the audio signal. An Audio button toggles between Wideband and Split modes, determining whether the attenuation affects the entire audio spectrum or only the high frequencies (based on the Frequency setting).

The Monitor button lets you toggle between listening to the audio output and listening to just the sidechain signal. Listening to the sidechain is helpful in identifying the troublesome frequencies in the source material. As you adjust the frequency value, the goal is to isolate what is heard in the sidechain to the most sibilant parts of the signal. Once you accomplish this, the side effects to the rest of the audio will be minimal.

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I've had great success running two *DeEssers* in series, one in Bandpass mode followed by another in Highpass for particularly sibilant vocalists. Several presets are tailored for male and female *sss* and *shhh* sounds. They're great starting points for dealing with your particular case of sibilance.

In addition, *DeEsser* works well on a variety of instruments, and even entire mixes. Any source that has peaky, high frequencies above 2 kHz can be tamed with *DeEsser*. You can't control the compression's envelope response time, but you don't need to. The response time is preset to be lightning fast, and it's exactly what you need in practically every application. Anyone who has fumbled with constructing a hardware de-essing chain will truly appreciate the simplicity and power of this plug-in.

#### THE NEW RENAISSANCE

As an antidote to the digitally precise nature of the original Waves compressor and EQ (C1 and Q10), the new Renaissance plug-ins emulate the nonlinear nature and desirable sonic char-

acteristics of top-level, well-respected analog hardware devices, while also providing simple user interfaces. After carefully studying many classic compressors and equalizers, Waves has created two plug-ins that not only simulate the sonic nature of the hardware that inspired them, but also stand on their own in terms of flexibility, ease of use, and musicality.

Renaissance Compressor (RC) is extremely flexible, yet its controls are quite simple, consisting of only five sliders and three on/off buttons (see Fig. 3). After first selecting the

mode, behavior, and character of the compressor with the buttons, you can adjust the Threshold, Ratio, Attack, and Release sliders to yield stunning results. And because RC is a plug-in, you can



FIG. 3: Renaissance Compressor combines terrific sound and impressive ease of use.

run multiple initiations (as many as your computer can handle) and apply each to a different track.

Renaissance Compressor is a soft-knee compressor/expander, meaning that

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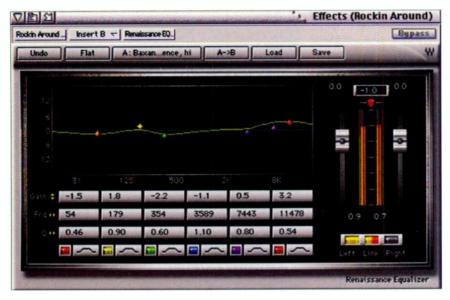


FIG. 4: Renaissance Equalizer is CPU intensive, but its stunning, analoglike performance is worth the extra horsepower.

attenuation actually begins 3 dB below the Threshold setting. The compression ratio can be anywhere from 1.01:1 to 50.0:1, and the expansion from 0.99:1 to 0.50:1. You set gain in 0.1 dB increments between +30 and -30 dB.

The Release control toggles between Manual and Auto Release Control (ARC). In Manual mode, you set the attack (0.5 to 500 ms) and release (5 ms to 5 sec). (The manual states the attack times incorrectly and omits the release times altogether.) In ARC mode, the proprietary auto-release function is program dependent, automatically choosing the optimal release setting. Both modes work well, but I found ARC to be more forgiving.

You can set RC to either Electro or Opto mode, which act in opposite ways. In Electro mode, the release gets faster as gain reduction approaches zero. When gain reduction goes above 3 dB, the release gets slower and the software acts more like a leveler. This results in increased average level and is ideal for loud, in-your-face signals, such as voice-overs, guitars, and aggressive mixes. In Opto mode, the reverse is true: as gain reduction approaches zero, the release gets slower. As more gain reduction is introduced, the release becomes faster, emulating classic opto-compressors.

RC's character can be Warm or Smooth. The Warm setting adds lowfrequency harmonics to the signal as more gain reduction is introduced. This is the second- or third-order harmonic distortion found in tube-based analog stages, which is usually perceived as "fat sweetness." You decide the amount of compression applied, but you don't get any other parameters with which to control the depth of coloration, and the effect is subtler than what you'd get with a real tube stage or something like an EL8 Distressor. However, the Warm setting does help solid-state signal paths that need a little muscle. The Smooth setting doesn't add any additional harmonics and sounds great if you're already using a good tube stage or have a killer signal path.

The Renaissance plug-ins have a distinctive look, and the metering on the compressor has a "classic" feel. Determining the attenuation is easy, thanks to the big yellow gain-reduction bar. The output section employs a brickwall limiter (similar to the limiter in Waves' L1 plug-in) that is preset to 0 dB. As the output exceeds 0 dB, a yellow light indicates limiting. The light grows brighter and eventually turns a burnt orange beyond 6 dB of limiting. Too much output limiting can cause digital distortion, but RC is certainly more forgiving than most digital compressors, giving it more of an analog feel and scoring high on overall "trust factor."

#### **REQ-REATION**

Renaissance Equalizer (REQ) is a 2-, 4-, or 6-band parametric EQ with an interactive graphic display (see Fig. 4). You set the Frequency and Gain settings by

simply dragging a display point for each band. (You can also click and drag numeric values or enter data with keystrokes—it's very flexible.)

The proprietary filters and bell curves are the result of painstaking research and design on the part of Waves. Based on the responses of some of the most highly regarded analog equalizers as well as breakthroughs in mathematical designs, REQ is quickly gaining a reputation among discerning engineers as one of the finest and most versatile EQs available. As with all Waves plug-ins, REQ comes with a detailed and informative electronic user manual (in PDF format). For more information about the design and philosophy behind REQ, visit the Waves Web site (www .waves.com).

Two of the basic design differences between REQ and most other digital equalizers (including Q10) are the shapes of the bell curves and filters. REQ typically uses a wider bandwidth when boosting and a narrower curve when cutting. The shelving filters are also variable rather than preset to a couple of curves. The distinctive effect of simultaneously boosting and cutting similar or identical bands (a popular trick on Pultec EQs) is also faithfully re-created. The end result is



#### **Native Power Pack Minimum System Requirements**

Mac: 603/120, 604e/180 with L2 cache for Steinberg's Cubase VST and other applications that support the VST plug-in architecture (G3 recommended); 32 MB RAM, 48 MB for multitrack applications; System 7.6.1

PC: Pentium/100, Pentium/166 for multitrack applications; 16 MB RAM (32 MB recommended); Windows 95/98/NT 4.0

a plug-in that sounds remarkably like a great analog equalizer.

REQ is a wonderful-sounding EQ. Practically any source in need of equalization can benefit from this plug-in. At times, the pure equalization capabilities of Q10 would probably work better, but I turn to REQ first for most of my digital audio EQ needs.

Both Renaissance plug-ins require more processing power than their Q10and C1 counterparts, but some sophisticated math processing is happening under the hood. This is why REQ is limited to six bands per initiation. If you can't make a sound happen with six bands, you can always open up another copy and run two REQs in series.

The only downside is that I usually run out of computer horsepower when I try to open too many initiations of the REQ plug-in. (On my system—a Macintosh G3/450 MHz with 192 MB of RAM-I usually have no problem running five or six REQs simultaneously with a few other miscellaneous plug-ins.) Of course, I can always create a new file of the equalized audio and free up the horsepower, which is what I usually do. It's easy to get spoiled and want to run 20 of these phenomenal plug-ins at a time. (I guess that's when it's time to remember how much 20 hardware EQs and cabling would cost.)

#### **WRAP IT UP**

All the plug-ins in Native Power Pack II—as well as the rest of the Waves collection-are superior pieces of software. The quality of the processing is outstanding and very musical. They are all immensely useful and easy to use. Anyone working with digital audio on a desktop computer needs these plug-ins.

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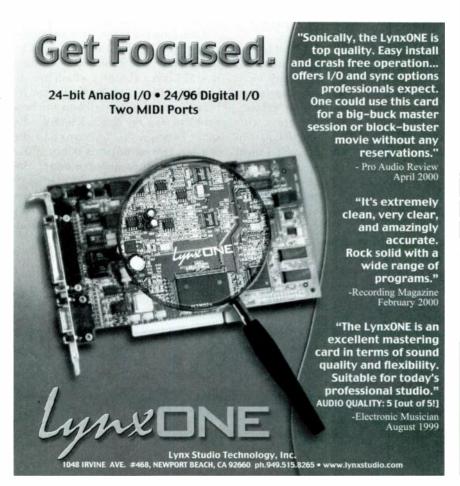




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# KURZWEIL

#### **EXPRESSIONMATE**

A ribbon controller for the hard-core MIDIot.

By Geary Yelton

spotlight focuses on the solo performance artist as she lifts what appears to be a graceful black sword that's roughly the length of a violin bow and cradles it in front of her. She holds it with one hand, while with the other she begins to play music by tapping and sliding her fingers across its surface. Scales, arpeggios, and rapid-fire riffs burst forth as she changes from one instrumental sound to another, her hands never leaving the black blade. You catch a clearer glimpse of her unusual instrument and see the name "Kurzweil" emblazoned across it. As she launches into a piano solo, people around you ask no one in particular, "What is that thing?" With a smile you tell them, "It's a Kurzweil ExpressionMate."

When Moog, ARP, and Buchla synthesizers were first introduced in the 1960s, some of their keyboards were supplemented by a ribbon controller. By sliding a finger along the ribbon, you could glide between notes with slide-guitar fluidity. But the ribbon controller had largely disappeared until Kurzweil introduced the K2500, which featured a ribbon controller with much greater capabilities than its forebears. Now the company's ExpressionMate adds a ribbon controller to any MIDI

synthesizer, along with a sophisticated MIDI processor and three independent arpeggiators that perform dozens of tricks never before possible.

#### **FIRST GLANCE**

The ExpressionMate comes with two parts and various accessories (such as cables and power supply). The first part is the ExpressionMate itself, a wedgeshaped black box with one small knob, a dozen switches, a dozen LEDs, and a 32-character LCD on the front panel (see Fig. 1). Various jacks grace the back panel (see Fig. 2). Cosmetically, it's a perfect match for any Kurzweil instrument and will look right at home sitting on your K2000 or PC88, but it works with any MIDI keyboard. If there's enough room, it can rest atop the synth's front panel. If that's not practical, you can mount the box on a microphone stand, using the included hardware.

The other part of the Expression-Mate is a metal-and-plastic ribbon controller 27.5 inches long by 1.5 inches wide (see Fig. 1). The ribbon is divided into thirds by white triangles that have been silk-screened onto it; other arrowheads indicate the half and quarter points within each third.

If you have the space, you can put the ribbon just above your keyboard by placing it on a foam strip or mounting it with Velcro (both of which are included). Otherwise, finding a convenient location for the ribbon can be a problem. On my Kawai K5000W, the space between the keyboard and the bottom row of buttons is not quite wide enough to accommodate the ExpressionMate's ribbon. I tried placing it on the overhang just below the front of the keyboard, but the ribbon's bulging ends prevented six keys from

being fully depressed. I had to settle for letting its edge hang over the top of the keyboard just a bit. Sometimes I just set the ribbon in my lap as I play. If you have trouble making it fit, use your imagination—I've even heard of someone mounting the ribbon on a sandeddown guitar fretboard and hanging it from a guitar strap.

A cable resembling a telephone cord is permanently attached to one end of the ribbon controller. By default, you place the controller so that the cable is on the right-hand end, but if you change the Ribbon Direction parameter, you can flip the controller around so that the cable is on the left.

On the back panel are two sets of MIDI ports: Main In, Out, and Thru and Aux In and Out. These let you control two independent MIDI devices and input data from two independent MIDI sources. You also get two jacks for footswitches, one of which can be a continuous pedal. In addition, there's an input that accommodates a breath controller, which lets you use this device with any synthesizer. Two buttons on the front panel labeled SW1 and SW2 work just like footswitches for whichever functions they're assigned to.

#### **OUTSIDE IN**

The ExpressionMate's greatest strength lies in its user programmability. It provides several physical controllers, including the ribbon, two panel switches, the breath-controller input, and the footswitch jacks. The panel switches and footswitches can be either momentary or toggle switches, depending on the current Setup (more on this shortly), and they can do anything from turning an arpeggio on and off to turning Sustain on and off.



FIG. 1: Kurzweil's ExpressionMate includes a control box and ribbon controller that you can use with any MIDI keyboard. You can mount the control box on the keyboard or a mic stand.

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"Really, it's an inspiration."

Willes for

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The ExpressionMate provides 64 user-programmable Setups, which define what the various controllers do. Each Setup gives the ribbon and other controllers an entirely new identity. Only the first 31 Setups are preprogrammed; these are called ROM Setups, but they can be edited, erased, or replaced by the user.

Setups are manually selected by turning the front-panel data knob or pressing the – and + buttons below it. If enabled, MIDI Program Changes can also change Setups. Three Setup Lists are accessible

with the < and > soft buttons. Press either button to go to the next Setup in the current List; press both buttons together to switch from one List to another. Setup Lists are useful for organizing Setups to make them quickly available during live performances.

You can assign the ribbon controller to control Pitch Bend, Volume, Effects Depth, Filter Swell, Panning, and just about anything that can be changed with a physical controller. When you slide a fingertip along the ribbon, the controller value changes in response. This response

can be modified with one of a number of controller curves (see Fig. 3). The ribbon can even send Note On messages; although the ribbon is not Velocity sensitive, the Velocity can be determined by a user-assignable controller.

The ribbon can be configured as one long controller or divided into three sections. If the Split parameter is set to Single, the entire length of the ribbon controls a single parameter. If Split is set to Three Sections, different controllers are assigned to each section of the ribbon. For example, one section can play notes, another can control Velocity, and the third can control vibrato. If you have a MIDI-controlled lighting console, these three sections could control fades for three different scenes. You could also use them with a sequencer to trigger start and stop, punch in and out, and tempo.

When the ribbon is split into three sections, it's easy to accidentally cross from one section to another and get a result you didn't intend. It would be better if some sort of ridge divided the three sections. This would be especially helpful on dark stages, and I don't think it would interfere with using the ribbon in Single mode.

The ExpressionMate has two sets of "virtual" controllers, each of which sends three preset values when you select a Setup. These are typically used to send Program Changes, initial MIDI Volume, and so on before you begin playing. They can also send strings of System Exclusive messages. In addition, there are three MIDI Remap controllers, which change incoming MIDI data into other types of MIDI data. For example, you can use the mod wheel to change MIDI Volume, or you can use Aftertouch to control Pitch Bend. You can even make the pitch-bend wheel work in reverse. Virtually anything can be reassigned to anything else.

#### **ARPEGGIATE THIS**

Besides the ribbon, the most interesting feature of the ExpressionMate is its interactive arpeggiator section, which offers three 16-step arpeggiators that can receive and send on separate MIDI channels. They can even be cascaded so that one sends note data to another. All three share a common clock, either internal or external, but they can all play with different rhythmic values, meters, and loop lengths.

The ExpressionMate's arpeggiators

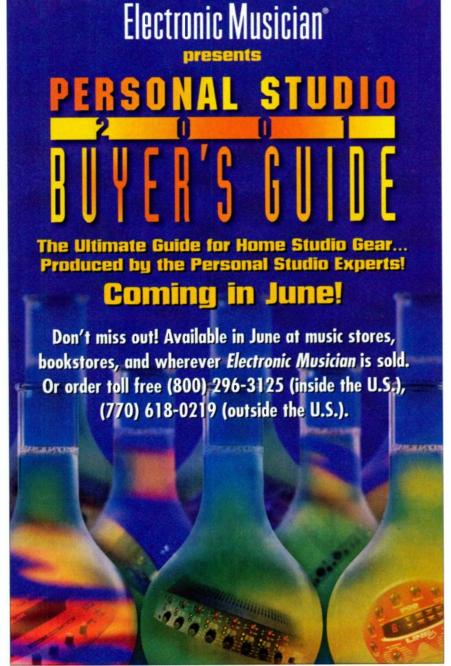




FIG. 2: The ExpressionMate's back panel includes a breath-controller input, two footswitch inputs, and two sets of MIDI ports.

are like others you might have used, but with a few new twists. In addition to the usual note orders (such as up and down, random, and the order in which notes are played on the keyboard), there are Random Walk, Fractal Walk, and Binary Walk, which are weighted variations on random order. Also, various latch modes determine how notes are triggered and whether they keep playing when you release the keys. For instance, turning on Glissando plays all the chromatic notes between the notes of the input chord. Besides playing all notes with an equal rhyth-

mic value, the arpeggiator can play at preset rhythms from a bank of eight ROM Rhythms or 64 User Rhythms.

Several Shift parameters transpose the arpeggiated notes with each successive repetition. For example, if you set the Shift Amount to a whole step up, the pitches rise a whole step each time the cycle plays until it reaches the maximum transposition you indicate with the Shift Limit parameter. When the cycle reaches this limit, it can stop, start over, reverse direction, or transpose itself, depending on the Limit Option setting.

#### SET 'EM UP

Most of the ROM Setups are intended for General MIDI instruments with the keyboard sending on channel 1. If you are controlling a non-GM instrument, you'll need to either turn off Program Change reception or edit the Setups. Kurzweil says the 31 ROM Setups are intended to show off the Expression-Mate's capabilities and encourage users to create their own Setups.

Some of the Setups are quite simple and straightforward. The best example is Setup 3, Big Strum. When you select it, the program on your GM synth changes to Jazz Guitar. When you play on the keyboard, nothing happens but when you slide your finger on the ribbon controller, the notes you play on the keyboard are triggered by the ribbon, all with equal Velocity. As you ascend the ribbon, the pitches are transposed up by octaves. In other words, the notes you play on the keyboard become a scale you can play on the ribbon. If you press all 11 notes in a chromatic scale, you can play every note on the ribbon, with a range of up to four octaves. If you press only the white



keys, the ribbon plays only white keys. If you play a chord, the ribbon plays only the notes in that chord, transposing up for each successive octave.

Big Strum, or a close variation, might be the only Setup you'll ever need, because it lets you play the Expression-Mate as a musical instrument, with all notes triggered on the ribbon. A variation on Big Strum could turn the device into a unique percussion controller.

Unfortunately, predicting exactly which notes will be sounded by the ribbon is sometimes difficult, and when you play a note it's easy to accidentally trigger an adjacent note. Kurzweil recommends that you avoid trying to play in two different thirds of the ribbon at the same time when in Single mode. The company also suggests playing no more than two notes at a time to avoid unpredictable pitches. This is a significant limitation that prevents the ExpressionMate's use as a generally polyphonic controller.

Other Setups are more complex. Setup 23 is a cool example: it summons the General MIDI program Marimba, but this Setup is inexplicably called C2Drms 3/4B. Playing the note C2 triggers Arpeggio 1 at a preset Velocity. Playing any note between C1 and C2 triggers Arpeggio 2, which plays at the velocity with which you press the key. Pressing

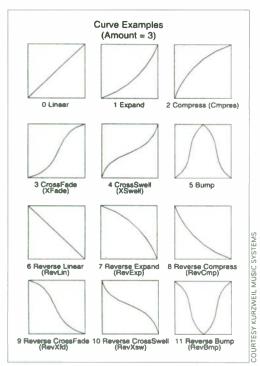


FIG. 3: A number of different controller curves let you tailor the ExpressionMate's response.

the SW2 button on the front panel stops Arpeggio 1, and pressing footswitch 2 turns off Arpeggio 2. Pressing SW1 stops both arpeggios. The ribbon's left-hand section controls reverb depth, the right section controls arpeggio tempo, and the middle plays a seven-note scale.

You must figure out most of this yourself, as the manual reveals only that playing C2 starts an arpeggio pattern and that the switches and pedals turn part of the pattern on and off. Most other ROM Setup descriptions are equally sketchy, so you'll have to explore them on your own to understand what they do.

One of the best ways to understand a Setup is by using a built-in utility program called *MIDIScope*, which works the same way as Kurzweil's *MIDIScope* for the Macintosh. MIDI data is shown on the LCD as it occurs; you can select whether you're monitoring data coming into or going out of any of the MIDI ports.

#### **MANUAL LABOR**

The 132-page user guide is full of information, but it's not organized in a top-down fashion. For example, it tells you how to program the Expression-Mate, while never revealing why you might want to do so, or where it might lead. There are no tutorials, and many concepts are poorly explained; you pretty much have to read the whole

thing just to grasp what's going on. An index is included, but the table of contents is a better place to find what you're looking for. In addition, the two "Quick Start" sections contain some erroneous and confusing information.

For example, the Arpeggiator "Quick Start" section begins: "1. Start with one of the Factory Setups. 2. Press the Arpeggiator button one or more times until the 'Z1' light above the Arpeggiator button comes on." This works with only a handful of ROM Setups, but the manual doesn't tell you that. With most Setups, you can press the Arpeggiator button until the cows come home, and the Z1 light will never turn on. Even when I found a Setup that worked in this manner, following the subsequent instructions didn't give me the described results-very frustrating if you're just starting out. I finally figured out that I



had to select one of the 31 initialized Setups (Fact Default) rather than a factory-programmed ROM Setup.

#### AN UNCOMMON EXPRESSION

The Kurzweil ExpressionMate is a complex device—perhaps too complex for many musicians. Most of you will have to dig in and do some programming before it will perform the right tricks, but you'll be rewarded with incredible flexibility. Unless you're content just using Setup 3 to play notes on the ribbon, it won't satisfy your need for instant gratification. Once you step outside the realm of General MIDI, most of the ROM Setups are of limited use. I wish the unit included more factory-programmed Setups and better explanations of what they do.

How useful you find the Expression-Mate depends largely on your knowledge of MIDI. If you're not into programming MIDI devices, you might be disappointed. However, if programming custom Setups is your bag, you'll have hours of fun with the ExpressionMate. The depth of its programmability and its sheer number of parameters will teach you a lot about MIDI, even if you're fairly knowledgeable already. Once you've learned your way around the device and created some useful Setups, you'll have an alternate controller like no other.

Geary Yelton has composed and recorded music for DuPont, Hitachi, Delta Air Lines, and various other advertising clients. He recently relocated from Atlanta to Charlotte, North Carolina.



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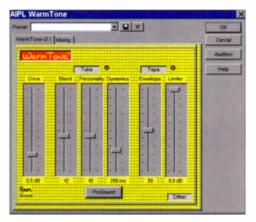


#### **AIPL**

WarmTone 2.1 (Win)

By Phil Darg

have a confession to make: I have never been fully satisfied with my digital recordings. When I compare them with mixes done on tape machines 15 years ago, they seem



With its simple yet efficient controls and excellent sound, AIPL's WarmTone is the ideal plug-in for producing tube-overdrive and tape-saturation effects.

to be missing something. Conventional wisdom attributes this to the "warmth" of analog recordings, which makes them smoother and easier on the ear. Digital recording is cleaner, but it's also colder and more sterile, and digital peaks (with their sudden precision) sometimes sound harsh.

AIPL's WarmTone DirectX plug-in (standard version, \$34; ProSound version, \$64) aims to solve that problem. This dual-purpose package adds both tube color and tape-saturation effects to tracks and mixes. The effects can be used separately or together under any compatible PC-audio host application.

#### **Easy Listening**

I tried WarmTone's tape-saturation feature first. The effect has only two settings: Envelope, which determines the degree of compression, and Limiter, which sets the threshold. Using even modest settings on

full digital mixes has a subtle but significant effect. Sharp attacks and transients become less strident. Harsh peaks are rounded into a gentle blending of instruments. Overall, the sound is indeed smoother, warmer, and easier to listen to. The effect works equally well on individual tracks.

WarmTone's tape-saturation emulation is clearly better than that of many other plug-in compressors. It produces few of the common sonic artifacts associated with conventional compression. The casual listener might not notice much difference, but those who strive for sonic quality will definitely hear an improvement. The effect is so pleasing that I used the tapesaturation feature on a number of my older mixes to improve their sound.

#### Tubular

WarmTone's tube-effect section adds various degrees of overdrive ranging from warm buzz to shredded metal. The adjustable sliders include Personality, which controls overdrive EQ bias; Blending, which controls the amount of overdrive added to the mix; and Dynamics, which adjusts rise and fall times. I used the tube-overdrive effect both alone and in conjunction with the tape-saturation effect and found that it works best with saturation. The results were some of the cleanest overdriven tube sounds I've ever heard.

Although the sounds were of high quality, I had trouble achieving sustained distortion. I would also prefer some additional controls (such as ad-

justable parametric EQ or a built-in amp simulator) to increase the range of sounds. Another drawback is that real-time performance requires a lot of CPU power, especially for stereo usage.

WarmTone is available only as a direct download from AIPL (www.aipl.com). The company has cut out packaging and shipping costs and passed the savings directly to consumers. This approach is both convenient and environmentally friendly. The file is a modest 600 KB, so it downloads quickly and easily. Users can try a free sample of both the standard and Pro-Sound versions. The ProSound version uses a higher bit rate for processing, and sounds sig-

nificantly better. It's well worth the price.

Overall, AIPL's WarmTone is a good addition to any serious musician's audio toolkit. Although its tube-overdrive effect lacks sufficient sustain, its tape-saturation emulation is superior; I'd buy the software for that feature alone. In fact, it has become my plug-in compressor of choice for mixing. If I were stranded on a desert island with only a few software plug-ins, I would definitely want AIPL's WarmTone to be among them.

Overall EM Rating (1 through 5): 4

#### CYBERWAVE EMS

Waveplant (Mac/Win)

By Jeff Obee

Sampler owners are on a never-ending quest for sensuous, otherworldly, magical sounds. Cyberwave EMS caught my attention with its *Waveplant* CD-ROM (\$45), "a collection of synthesizer sounds," the manufacturer claims, "that have never been heard before."

The Waveplant is a hybrid hardware-software synthesizer. The software runs on a PC and connects to the hardware via ISA ports. It controls the hardware components: three standard LFOs, nine oscillators (saw, square, variable pulse, triangle, noise, and four simple additive waveforms), and nine ADSR envelope generators. The filters are separate hardware modules. They consist of three 24 dB/octave, variable-state voltage-controlled



The Waveplant sample CD-ROM from Cyberwave EMS captures the modular sounds of the Waveplant synthesizer, a hardware-software hybrid in which the hardware handles sound generation and the PC software controls the hardware and provides modulation matrices.

filters (VCFs) and three 12 dB/octave VCFs.

The hardware handles sound generation; the software provides program storage, parameter adjustment, key scanning, and modulation matrices. The modulators are composed of 36 independent LFOs arranged in groups of four matrices. They are used to create phase shifts, pan effects, chorusing, and so on.

#### Critical Issues

Waveplant has some conceptual short-comings. The disc offers no multisamples—a great disappointment. It has categories for Organs, Leads, and Polysynths, but without at least two samples per octave, you won't get enough of a range to create those types of patches. The booklet states that Waveplant contains 120 MB of samples; when you consider that 650 MB can fit on a disc, there seems to be no excuse for omitting multisamples.

The booklet includes a list of categorized sample types; unfortunately, it's not in the same order as the audio on the disc. Furthermore, the sounds themselves aren't individually listed or indexed. (According to Cyberwave EMS, the documentation has been revised since I reviewed the CD-ROM.)

Waveplant's sounds are in three file formats: audio, WAV, and SoundFont. Cyberwave EMS recommends using the CD-ROM with a SoundFont-compatible system. As a Macintosh user, I'm unfamiliar with SoundFonts, so I didn't review them, but I did test out Waveplant's WAV files. They work fine for one-shot samples, and you can view them on your computer, eliminating any doubt about what you're hearing.

#### **Wave Sounds**

Waveplant is a collection of individual sound effects rather than a set of traditional synth sounds. The first few times I listened to the disc, I got the distinct impression I was listening to a modular synth—standard waveforms with varying degrees of resonance, sample and hold textures, filter sweeps, and the like.

I found some appealing sounds. The numerous modulators give movement to many of the sounds, with rich, emerging overtones and subtle granular effects. Some files are quite large, progressing through changing segments that you can snip and use as shorter bits. The clangorous and ring-modulated samples are good, too.

The sounds in the Otherworldly category, however, aren't quite alien enough for my tastes. Tonally, these samples have a limited, analog quality. I would like to

see Cyberwave EMS more fully explore the Waveplant synth's capacity for creating ambient textures.

Waveplant's bass samples aren't my cup of tea either. Although my lack of familiarity with SoundFont and the absence of useful indexing prohibited a complete analysis, I nonetheless found the distorted bass samples too long: distortion occurs at the beginning of the sound, followed by a severe drop in volume and a smooth decay. But if you find a suitable loop, some of the bass samples might work well as synthlike drones.

#### **The Final Wave**

After reviewing Waveplant, I found myself asking, what if? What if the sounds were offered as multisamples? What if the categories were focused, offering distinct, biting leads; broad, ambient pads; a more diverse palette of effects; and tight, concentrated bass samples? What if the documentation and indexing provided a better road map of the available samples?

I hope that Cyberwave EMS will take these questions into consideration when it creates the next Waveplant-synthesizer sample CD-ROM, because *Waveplant* has potential. If you're looking for single-shot modular effects, you may find this disc to be well worth its low price.

Overall EM Rating (1 through 5): 2

#### **NEMESYS**

GigaHarp and Upright Acoustic Bass (GigaSampler)
By Zack Price

NemeSys's GigaHarp and Upright Acoustic Bass (\$299 each) are CD-ROMs in Giga-Sampler format that faithfully reproduce the natural sound of the harp and upright acoustic bass. The company accomplished this by sampling each instrument's individual notes in 24-bit stereo at multiple dynamic levels for the entire length of their natural decay. Extensive use of keyboard mapping and MIDI Control Change messages generates expressive real-time nuances for each instrument. The accompanying help files detail each program's parameters and how they relate to real performance techniques.

#### **Zing Went the Strings**

GigaHarp contains samples of every string of a pedal harp at four dynamic levels (soft, medium-soft, medium, and loud finger plucks); open-string harmonics at two



PICKS ynamic levels; and

dynamic levels; and fingernail-picked samples at one dynamic level. (The four levels of finger plucks are Velocity switched; the mod wheel accesses and controls the harmonics and fingernail-picked notes.) These samples are mapped to five patches. Harp 1 and Harp 2 differ only in their Velocity sensitivity: Harp 1 requires a wider Velocity range to affect patch dynamics; Harp 2 is more sensitive. Damped Melodic Harp has shorter programmed release times, allowing you to damp notes with Note Off and Sustain Pedal messages.

The other two patches (Glissando Harp 1 and 2) make it possible to play realistic harp glissandos in tempo and in the proper key signature by performing thumb and finger slides on the keyboard's white keys. On a real harp, which is a diatonically tuned instrument, you change the key by depressing and locking combinations of seven pedals—one for each note in the diatonic scale—with each pedal combination lowering or raising the natural notes a half step. GigaHarp emulates this technique by using Data Entry sliders as virtual harp pedals. For more creative possibilities, the disc offers a virtual pedal position (double sharp) not found on real harps.

#### **Upright and Upstanding**

Upright Acoustic Bass is a plucked-bass library, so it has no bowed samples. However, it does offer multilevel samples of picked notes, slides, muted string sounds (x-notes), and picked harmonics. These samples are mapped out over an 88-note keyboard, and in some patches the unassigned keys toggle between different sampled articulations. The disc also features body-resonance samples that play as the notes release. You can control the amount of body resonance in real time using the mod wheel.

Foot Controller messages (MIDI CC 04) toggle between ascending and descending slide samples; Breath Controller messages (MIDI CC 02) switch between fast and slow slide samples and their target notes. Within the keymap of B1 to E4, the sustain pedal switches between sustained note samples and resonant muted notes; from F4 to D5 it switches between sustained notes and harmonic slides.

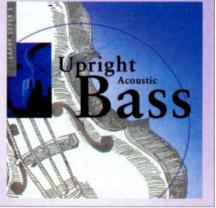
There are seven different patches, each with its own set of samples. The first patch, Optimal Upright Bass, employs carefully selected sustain samples that are equalized to better cut through a mix. The 12-Position Bass patch uses unprocessed and naturally decayed samples that are mapped chromatically according to position for a darker, more mellow sound.

Few bassists choose to play entire lines on one string, and the same note played on different strings will vary in timbre. With this in mind, the Switchable Position Bass patch plays notes from the proper individual string to emulate the string choices favored by bass players. Keys F, G, A, and B1 have no assigned samples; playing these keys bypasses the switch to the new string and extends the range of the current string. The last four patches contain multisamples of every note on each string, and individual strings are assigned to separate MIDI channels for use with MIDI guitar and bass controllers.

#### Do One Thing Right

With roughly half a gigabyte of samples apiece, GigaHarp and Upright Acoustic Bass have everything you need to faithfully reproduce the sounds of their namesake instruments. Control parameters are well thought out, making real-time control uncomplicated. Considering the quality of these CD-ROMs, each is well worth





NemeSys's GigaHarp and Upright Acoustic Bass CD-ROMs will help you create expressive and authentic-sounding harp and bass performances.

the price. When it comes to creating realistic harp and bass performances, Giga-Harp and Upright Acoustic Bass are as good as it gets.

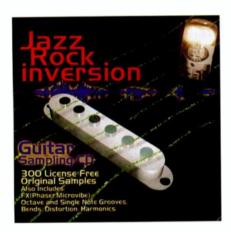
Overall EM Rating (1 through 5): 5

#### **GROOVEADEMICS**

Jazz Rock Inversion

By Jeff Obee

Guitar sample CDs come in two basic types. One features guitars that are multisampled and programmed in detail for "true to life" playability via keyboard or guitar MIDI controller. The other emphasizes licks, riffs, and chordal comping



GrooveAdemics' Jazz Rock Inversion guitar sample CD offers a complete range of mellow jazz-chord inversions.

performed by a guitarist and sampled as short chunks. GrooveAdemics' Jazz Rock Inversion (\$19.99, audio CD) falls into the latter category, though it makes a limited foray into the former as well.

#### **Inverted Guitar**

The first third of the disc focuses on jazz chords, comped on a Howard Roberts semihollow-body jazz guitar. The guitar sounds quite good; the samples, however, are played only at medium-soft velocities, severely limiting the possibility of dynamic performance. Frankly, these samples made me snooze. You get a lot of chords, though, and according to the manufacturer, that's what this disc is really about. The chords range from dominant ninths to major, minor, and dominant seventh inversions. If putting together a mellow jazz performance is your thing, these samples will definitely do the trick. Picked and

thumbed samples of open strings round off this section of Jazz Rock Inversion. Unfortunately, they don't adequately represent the range of the guitar and are therefore of limited use.

The next section, featuring a '79 Strat playing various short phrases and licks, fares better. It includes single-note mutes, octaves, octave-to-single notes, up and down slides, slide-mutes, pull-offs, and so on. These are nicely played and can be used as "nibbles" in many contexts. I was confused, however, by the inclusion of a few chords—A7; D sus4; C, G, and A major; and E and A minor—that are single strummed with no variations or rhythm patterns (the one strummed variation is a brief, uninspiring C-major-to-E-minor chord progression). These chords hang limply and have no real application.

The same goes for the few blues licks, and I do mean few. Track 75 has two slowblues chord comps and an ending line in the key of A, but no corresponding samples in the keys of D and E to round out a 1-4-5 blues progression. Track 76 has the only other blues parts, and although it actually has a 1-4-5 progression, the 4 and 5 chords are merely rhythmic slides, not the main comping pattern set up in the root chord. There are hardly enough samples to put together a workable blues line.

The last groups of samples are chord strums played through effects such as a Voodoo Micro-Vibe tremolo and a Maestro Phase Shifter. Like the Strat strums, these chords lack rhythmic variations and go nowhere. The distortion pieces, rock bends, and licks at the end of Jazz Rock Inversion might work as musical incidents, but overall the disc lacks idiomatic variation.

#### **Licks and Lumps**

GrooveAdemics can't seem to decide what sort of CD Jazz Rock Inversion should be. Give me a disc of jazz guitar with lots of dynamics and phrases; one with Strat samples that key on blues and funk bits or blazing fusion performances; or one of rock and alternative chords and strumming parts. As it is, you don't get enough of any one approach to make this disc worthwhile. The audio quality is good, and the CD is very inexpensive, but the playing is merely acceptable. Jazz Rock Inversion left my hopes for a collection of Scott Henderson—style jazz-rock guitar samples woefully unfulfilled.

Overall EM Rating (1 through 5): 2

#### **BIG FISH AUDIO**

Roots of India

By Dan Phillips

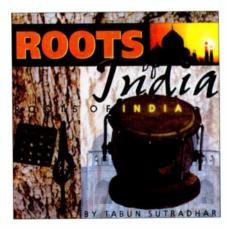
Roots of India is Big Fish Audio's collection of Indian percussion loops (\$99.95 for the audio CD; \$199 for the audio CD and Akai CD-ROM, which should be available by the time you read this). I know very little about Indian music. My introduction to the genre was the Beatles' "Within You Without You." Other than that, I've only listened to a few Ravi Shankar discs and gone to some live shows.

So instead of embarking on a high-minded ethnomusicology adventure, I approached *Roots of India* with a more mundane (and practical) purpose: to apply some of the country's ancient, highly evolved percussion timbres and rhythms to contemporary Western pop music and film scoring.

#### **Back to the Roots**

Happily, Roots of India fits perfectly into a Western studio. The performances are extremely tight, and the tracks are impeccably recorded. The producer, Tabun Sutradhar, has included instruments and rhythms from across the Indian subcontinent, giving the disc a lot of variety. The selections cover a wide range of timbres and moods, from the mesmerizing Bhajani Naman pattern at 80 bpm—with its deep, slow drums and floating finger cymbals—to the riotous, 145 bpm Bhangra, a Punjabi rhythm pattern that features clangorous drums and driving tambourines.

The Southern Indian loops are among my favorites. Several of them move at a cool and slow 100 bpm, with a jaw harp adding a twangy counterpoint to the steady



Big Fish Audio's Roots of India serves up rhythms and percussion sounds found throughout the Indian subcontinent.

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16th-note drum patterns—the kind of sound that incites a smooth, swaying dance. I also liked the Gujarati Garba patterns, with their strict four-beat feel and heavy downbeat, as well as the hypnotic, syncopated Rajastahani rhythms.

#### **Break It Down**

The loops are well organized, grouped by region or musical tradition. Each group contains both signature rhythms and instrumental ensembles. Most rhythms are provided in several different versions with identical tempos, so mixing and matching them is easy.

The rhythms are typically built from two or more interlocking drum patterns. Each pattern involves at least one low-pitched drum and one high-pitched drum along with one or more percussion tracks, such as finger cymbals or tambourine. Nearly all phrases include a main mix and two to four solo-instrument or submix versions, allowing access to the individual components.

The loop lengths are generous, often four measures long. I wish, however, that the loops ended on the downbeat. Most of them end on the upbeat, which makes looping a little more difficult.

The disc's documentation is solid and free of errors. It includes a helpful glossary that describes some of the percussion instruments featured in this collection. It would be even better, however, if it described all the percussion instruments, as well as each rhythm pattern.

#### A Taste of India

All in all, this disc took me on a very successful audio outing to the Indian subcontinent. If you want to introduce an exotic percussion flavor to your popmusic tracks, I highly recommend giving Roots of India a spin.

Overall EM Rating (1 through 5): 4.5

#### ANTARES

Auto-Tune 1.6 (Mac/Win)

By Phil Darg

How many times have you laid down a great track, only to discover that it's out of tune and useless for the final mix? Time for another take, right?

Not anymore. With Antares's Auto-Tune plug-in (from \$299-\$599, depending on the host platform), you can correct the intonation problems of mono audio tracks

after they've been recorded. The plug-in is available in several formats: DirectX, VST (Macintosh only), TDM, MAS, and, by the time you read this, RTAS. I reviewed the DirectX version. Antares also offers a dedicated Mac program called AudioStream that runs Auto-Tune as a stand-alone application.

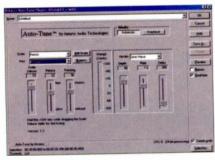
#### Tune In

The official pitch (no pun intended) is that Auto-Tune automatically corrects out-of-tune notes without introducing any artifacts or otherwise altering the sound. You can constrain your melodies to different scales and keys, and you can even delete notes from the provided scales to limit the pitches on a track.

Auto-Tune offers two modes: Automatic and Graphic. Automatic mode lets you simply choose a scale and apply the effect to a track. You can use major, minor, and chromatic scales, as well as a host of ancient and modern tunings. In addition, you can easily create your own note collections and save them as presets.

In Automatic mode, the Retune slider controls the rate at which pitch correction is applied. With a fast setting, for example, notes are "snapped" to the nearest entry in the scale that you are correcting to. This quantizes the pitch and can produce effects such as those found in Cher's "Believe." (In fact, Auto-Tune is one of the tools used to produce the effects in that song.) As you move the slider toward the opposite extreme, an increasing amount of the natural portamento is retained in the track; at the slowest setting, there's no correction at all. You can also use the Vibrato option to add pitch variations of your own (they can follow a sine, ramp, or square curvature).

Automatic mode offers many useful parameters to tweak. Nonetheless, you can get good results just by using the default



Antares's Auto-Tune plug-in automatically corrects out-of-tune notes. You choose between Automatic mode and a Graphic mode that provides a pitch-correction drawing function.

values and setting the scale to "chromatic."

In Graphic mode, Auto-Tune reads a selected track and displays its pitch graphically. Using functions such as line- and curve-drawing tools, you can precisely trace the tuning action that Auto-Tune will apply to the selected audio. Your audio is displayed as a waveform, so you can easily see where the changes will occur. Although Graphic mode is more complicated than Automatic mode, it offers exacting control over the tuning and enables you to create interesting pitch shifts and other effects.

#### Tune Up

How well does Auto-Tune perform? I tested the software on several electric-guitar, vocal, and synth tracks. In general, I was impressed with its ability to tune the tracks smoothly without adding unwanted artifacts. Auto-Tune works best on sustained notes, but with a little effort it can be useful in rapid note passages, too. The large number of scales, which include microtonal offerings as well as standard Western pitch collections, is a bonus.

The only unpleasant effect that I noted was an occasional pitch-sliding sound, which I was able to correct to some extent by adjusting the Tracking control. Running the processed audio through an external tuner, I found that *Auto-Tune* has virtually perfect pitch. In the few cases where the program wasn't initially effective, I improved its performance by editing the undesired notes out of the scale.

#### **Tune Out**

Overall, Antares's Auto-Tune is a great tool that improves the intonation of mono tracks without compromising sound quality and reduces the number of takes needed for a session. Although learning to apply the effects correctly takes time, using Auto-Tune has obvious benefits; Antares should be praised for creating a truly unique product.

I have some minor quibbles: my version occasionally gave me incomplete onscreen text, and the current release ships on floppy disk. (According to the company, all future releases will be on CD-ROM.) But I have no major problems with the software. Auto-Tune is more expensive than some PC plug-ins, but Mac users will find the price pretty typical. In any event, the hours it will save you easily justify the cost.

Overall EM Rating (1 through 5): 4

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Lee Stein, Western Regional Sales Manager, Sennheiser

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#### **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-hit audio at 44.1 or 48 kHz, or 12 tracks at 88.2 or 96 kHz. Up to 24 tracks can b 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 TI format allows compatible software to recognize virtual tracks without have to load, reposition and trim each digital file.

#### Transport Controls

- MIDI In Out, and Thru ports are built-in for MIDI

#### Editino-

- · Built-in editing capabilities include cut, copy, paste solit and ripple or overwrite
- · Supports destructive loop recording and nondestructive 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
- Can generate or chase SMPTE timecode or MIDI Time Code
- · Word Clock In. Out, and Thru ports

#### 1/O Ontions-

- 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/DtF 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

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 ezsy via a friendly user interface. FlexBus and EZ routing capabilities as well as a large informative LCD and ultra-fast short cut keys



VM Basic 72

Digital Mixing System

#### Features-

- 94 channels of digital automated mixing (fully expanded)
   Up to 48 channels of ADAT/Tascam T-DIF digital audio
- /O with optional expansion boards and interfaces
- Separate console/processor design
- Duiet motorized faders, transport controls, total recall parameters including input gain, onboard mixer dynamic automation and scene memory
- 24 fader groups, dual-channel delays, 4-band
- parametric channel EQ + channel HPF
   FlexBus and "virtual patchbay" for unpara leled routing flexibility

#### 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 mong effects
- 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 24bit digital I/O, totalling 24 I/O per expansion floard

- Up to 16 stereo (or 32 mono) multi-effects processors using optional VS8F-2 Effects Expansion Boards (2 stereo effects processers standard)
- COSM Speaker Modeling and Mic Simulation technology
- 5 1 Surrounc mixing capabilities
- FZ 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.
- · DIF-AT Interlace Box for ADAT/Tascam -- Converts signal: between R-Bus (VM-24E expansion board required) and ADAT/Tascam T-DIF. Handles 8-in/8-but digital audio. 1/3 rackmount size.

  • VM-24C Cascade Kit — Connects two VM-Series
- proce-sor units. Using two VM-7200 processors cascaled and fully expanded with R-Bus I/O, 94 channels of audio processing are available.

#### 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 commercia 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 be another recorder or even a DAT or CD Truggarder to allows pre-initial digui-bouncing within a single unit, or externally to another recorder or even a DAT or CD Truggarder. Up to 16 DTRS machines can be synchronized together for simultaneous, sample accurate control of 128 tracks of digital audio.

MICROBOARD StartREC Digital Audio Editing/CD Duplication System

- · Selectable 16 bit or 24 bit High Resolution aud o
- · 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.
- rd 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 SiPDIF

#### The MPX 500 is a true stereo 24-bit dual-channel processor and like the MPX100 is powered by Lexicon's proprietary exichip and offers dua-channel processing. However, the MPX 500 offers even greater control over effects parameters, has digital inputs and outputs as well as a large graphic : display.

MPX-500 24-Bit Dual Channel Effects Processor

- · 240 presets with classic true stereo reverb programs as well as Tremolo Rotary, Chorus, Flange, Pitch, Detune 5.5 second Oxlav and Echo
- Balanced analog and S/PDIF digital I/O
- · 4 decicated 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
- · 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 Deray
- · Absolute Repeat Control
- · Up to 10 seconds of Delay . 50 Factory/100 User presets
- S/PDIF digital I/O, 44.1-48kHz · Balanced 1/4° Jacks - Dual I/O
  - 24 bit internal processing

24 bit A/D-D/A converters



- 2X, 4X, or 8X recording speeds
- . 6 2GB IDE hard drive

run CD-R duplication.

 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

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

- · Coaxial SP/DIF or AES/EBU digital input plus optica
- XI R halanced 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



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#### NT-2 Condenser Mic

The RODE 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 RODE 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
- · High pass filter switch
- · Omni and cardioid polar patterns
- · -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





#### KSM-32SL Cardioid Condenser Mic

reviews are raving about Shure's new "classic" microphone. The KSM32 features The reviews are raving about Shure's new "classic" micropnone. The Namas really Class A. transformeriess preamplifier circuitry, low self-noise and increased dynamic range, all necessary for critical studio recording. It has a 15 dB attenuation to the Company of suitable for a variety of sound sources. syntamic range, an incleasary for clinical studies for a variety of sound sources switch for handling high SPLs, making it suitable for a variety of sound sources including vocals, acoustic instruments, ensembles and overhead miking of deums 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 five applications, the KSM32/CG has a charcoal grey finish and includes a swivel mount and padded zipper bag.

• Frequency response 20Hz - 20kHz



#### C4000B Electret Condenser Mic

his 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
- · High Sensitivity

- · Extremely low self-noise





#### 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



#### **AT4047SV Cardioid Condenser Mic**

he AT4047 is the latest 40 Series large diaphragm The AT4047 is the latest 40 Series large mapringing condenser mic from Audio Technica. It has the low self noise, wide dynamic range and high sound pressure level to the condition studies and sound 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

#### ERGENCE 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 a 1M
- Peak Acoustic Output 117dB SPL (100ms pink noise at 1Mt
- XLR outputs from power amp to speakers
- · Matched impedance output cables included

#### Amplifier

- · Amplifier Power 250W (continuous rms/ch), 400W (100ms peak)
- · XLR, TRS input connectors · Headphone output
- . 5-position input sensitivity switch with settings





- . -6dB LF Cutoff 40Hz
- . 5 position wall proximity control
- 5 position listening proximity control between near mid and far-fie d monitoring
- Power, Overload; SPL Output, Line VAC and Output device temperature display.

- 2-way acquistic suspension with a # 5-in/h treated paper woofer and a 1-inch aluminum dome tweeter
- . Fully magnet cally Shielded with an 18-inch



#### 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 mineralfilled polypropylene cone with 1-inch diameter high-temperature voice coil and damped rubber surround LF Driver Magnetically shielded 25mm diameter
- ferrofluid-coo ed 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 Combination Power On/Clin LED indicator



# 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 frequenty and 120 Watt low frequency amplification HF adjust +1dB, Flat, -1d8
- LF adjust -3dB at 45, 50 and 65 Hz

#### TRM-6 Bi-Amplified Near Field Studio Monitors

Offering honest, consistent sound from top to bottom, the TRM-6 bi-amplified studio mointors are the ideal reference monitors for any recording environment whether tracking, mixing or mastering. Supported by Haffer's legendary amplifier technology that provides a wide and accurate sound field, in width, height and also depth.

- 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 reszonse ±2dB
- 75 Watt HF, 150 Watt LF amplification





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THEY ALL FEATURE - • Mac CS and Windows compatible ludes software drivers for compatiblify with all in foday's popular oftware plus AudoLask MOTUs samp accurate audio workstation software for Mm (S • Host computer determines the umber of tracks that the software can record and our same on our as well as the arount of mothers effects processing 4 can support integral metering for all inputs and outcuts

· AudioDesk Audio Workstation Software for Mac OS features 24bit recording multi-channel vieweform editing, automated virtual musing graphic editing of ramp automation real-time effects plug ins with 32 bit floating point pro-essing crossfades, support for thirdparty audio plug ins (in the MOTU Audio System and Adobe Premiere formats), background process in a file out of operations sample-accurate editing and placement of aurio. And more



2408 well FFATURES-- 7 banks of 8 channel UC 1 bank of analog 3 banks of ADAT optical 3 banks of Tascam TDIF plus staren \$1/PDIF • Custom VLSI chip for O capabilities . . Format conversion between ADAT and Da-88

 8x 24-bit 1/4 balunced 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 - 21-bit analog and o interface • State-of-the-art 24-bit A'D'A • Simultaneously record and play back 8 hannels of balanced (TRS) +4 dB audio .

metering for all injurs and outputs. Headphone jack with volume knob



308 Features - • 8 channels of coaxial S PDIF using . RCA input and 4 RCA output connectors • 8 channels of optical S PD.F using 4 toslink input and 4 toslink output

connectors • 8 channells of AES EBU using 4 XLR male and 4 XLR female connectors • Word Clock | O allows the 308 to synchronize with digital audio environments



24i Features - • 24 high quality, 24-bit analog inputs . Balanced 1/4 asalog outputs . Optical and coavial S/PDIF outputs . Front panel headphone output

with level control . Word Clack 1.0 . Connect up to three 24i rack I/Os to a PCI-324 audio card for a total of 72 inputs and six outs

#### digidesign



#### DIGIOO1 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 environment for the Mac and Pro. The Util-101 orders a 24-bit multi-I/O breakout interface along with Pro Tools LE software—based or 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 plugins. 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

proce sing- Real-Time AudioSuite (RTAS) is a hostbased 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 ED: Oynamics II- compressor. limiter, gate and expander/gate, Mod Delay - short slap, medium, and long delays with modu ation capabilities for chorus or flange effects and dither AudioSuite o ug-ins include Time Compression/Expans on, Pitch Shift, Normalize, Reverse

#### MIDI Functions

- MIDI functions are use graphic controller editing, plano roll display up to 128 MIDI tracks and editing options like quantization transpose, split notes, change velocity and charge duration
- . MIDI data can be edited on the fly

Also available with MOTU's award-wirning Digital Performer audio sequencer software package



#### Digital Performer 2 MIDI/AUDIO Software for Mac

Digital Performer is an integrated multitruck digital audio and MIDI Digital Perormer is an integrated multitriack orginal about and integrated multitriack orginal about and integrated with advanced tools for a wide variety of audio applications. Sample accurate editing, loop base audio capture, realtime ESP effects and the best MID timing/ esolution available insures unlimited creative potential

· Includes over 50 real-time M'D) and audio effects plugins • POLAR window - which provides Interactive audio loop recoroing • 24-bit recording and editing • 32-bit native effects processing incredible sounding EQ and other FX • 64-bit Master Yorks Limcer and Multiband Compressor plug-ins included • Sample-accurate - the most reliable waveform Editing and Eightest sync you can get • Samulers window - drag & drop samples between your Mac and your Sampler • PureDSP stereo pitchshifting and time-stretching. Unlimited audio tracks, real-time editing, full automation and remote control. Quel-Time digital vides support

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#### NEW FEATURES-

- · Full Plug-In FX automation and increased 3rd party Plug-in support
- Drum Edito
- Adjustable Display Resolution from 2 to 10,000 PPO. Tick alues up to four decimal places can be set allowing 1000 times greater ed ting resolution. For example, if you are used to editing MIDI data at 480 PPQ, you can set your edit resolution to 48% COO or 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

SPARK 1.5 2-Track Editing For Mac

#### AMM-1 Microphone Modeler The AMM-1 Microphone Modeler uses

ANTARES patented technology to create precise digital models of a wide variety of microphones, from historical classics to modern exotics and even imdustrystandard workhorses. Simply tell the Microphone Modeler what microphone you are actually using and what microphone you d like it to sound lie. It is as simple as that Available as a plug-in for the TDM and MAS environments. with DirectX and Mac VST not far behind



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paring 1 • Du lys are adjustable in milliseconds and note values • Tap but delay times or pattern in a the Tap Pad MetaTanger Vintage tape-flaming phaser-emulation and special effects • True dual-dual transparency vintage tape-flaming phaser-emulation and special effects • True dual-dual • Wet signal include filters so you can flampe or plase just part of the special • Factor preacts of wintage emulations (Mutron MRX flichycoo Park) and more MondoMod • AN • FM. and Rotation (stereo panning) modulators • Ge tte wandering guitar sis in princing or bizarre destructive effects • Single LEO drives all modulators with individual programment of the solitage of the special programment of the spe



- Spark is professional 2-track audio editing software for the Power Macintosii 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 movia, edited and then exported along with the video to a new file. Bundled with Adapted's Toast so you can burn your audio directly to CD. · Browser View- File database audio editor and play ist 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 ron-destructive regions Supports AIFF Sound Designer WAV

and QuickTime file formats

DSP Processing Includes- Normalize. Reverse Fades, Crossfades, and Sample Rate conversion and realtime

#### Tima Stretching

ile menu

- 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 an burn your audio on CD Extract audio from a quicklime 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



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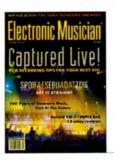
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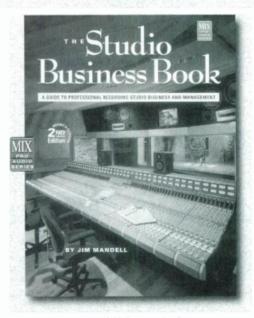
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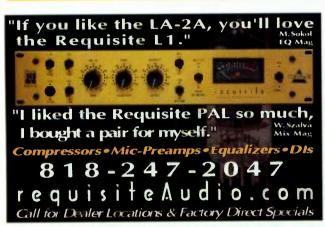


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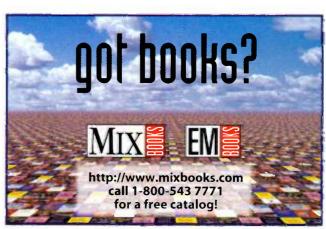
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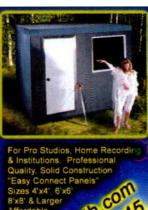
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# FINAL

# DUI, GUI, FUI!

sage once said, "A person has control over nothing, only varying degrees of influence over things." All right, so I said it, but I think that it's true, whether we're talking about the course of one's life or about one's interaction with machines. Anyone who has had the experience of hydroplaning in a car realizes that a steering wheel, brakes, and an accelerator do not add up to absolute control.

Readers brave enough to have subjected themselves to my mutterings over the years know that I have many times reviewed, reported on, and opined about the joysticks, mixing surfaces, and other devices commonly referred to as controllers. (Maybe we should call them influencers. Maybe not.) Well, here I am, on that subject again—as I surely will be in the future—because all the digital gewgaws in the world don't amount to a hill of silicon without a facile means of accessing their power.

Currently, the techniques of control (a word I'll use because it is the most common term for the concept and it's two letters shorter than *influence*) are in an interesting transitional phase. While software and hardware devices continue to quickly grow in sophistication and complexity, methods of controlling them are huffing and puffing to keep up—and not making it.

Take the current crop of control surfaces and digital mixers. They have lots of knobs, sliders, and buttons—things that proved to be well-suited for controlling machines long before electricity—yet many of these devices still leave us wallowing in a morass of menumining, masking-tape strips, and scribbled scraps of paper. Although they offer superior gestural control, these surfaces plainly don't provide enough information about the state of things.

With an assignable control surface, it's easy to become confused about what a control is doing at any given time. High-end, large-format, programmable mixing consoles often feature "soft scribble strips" that identify

what a fader or control is doing. Cost and real estate make that kind of display prohibitive for hardware sold to mortals; in fact, even the consoles that have it can usually display only eight characters or so per fader.

Knobs, faders, and buttons are great for varying parameter values but aren't so good for navigating through menus. Neither are keyboards and mice, in my opinion. I don't know about you, but I'm sick of constant mousing and clicking to get things done, and so is my ulnar nerve.

So what would be better, and will we ever see it? Let's look at the two problems I

First, information display. Flat-screen monitors are becoming increasingly available at relatively affordable prices. This is a good direction to go in, and I hope (and think) that these will continue to get larger, thinner, and cheaper. When they do, it will be possible for your display to be almost any large surface, such as a tabletop. I think that people can deal with seeing large amounts of information if it's not all tiny and crowded together.

mentioned and the technologies that I think can help.

Next we add touch capability to those monitors. Last year I had the opportunity to review a mixing console that used TFT touchscreens as the primary means of accessing the software, and I was floored by how fast and intuitive it was to use. Navigating down three or four menu levels took practically no effort at all.

Finally, I eagerly await the day that voice control matures. Even systems that recognize only a limited number of commands—say 50 to 100—could make controlling the primary functions of sophisticated systems fast and easy. This is a tough technology to develop, but industries larger than ours have a real need for it, so I think it will happen.

I would truly be delighted to trash my mouse and use my keyboard only for entering text, doing the rest of my control with sliders, knobs, buttons, touchscreens, and my own dulcet tones. Maybe I'm just Dreaming Under the Influence here, but I don't think so.



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