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I N S I

FEATURES

Cover Illustration by Dave Ember

DEPARTMENTS

WRH

COLUMNS

94

40

REVIEWS

WRH

The Tortoise and the Hare

In Aesop's fable "The Tortoise and the Hare," the plodding tortoise outraces the speedy hare because the former is steady, hardworking, and indefatigable, whereas the latter is arrogant and takes his ease, certain that victory is inevitable. The hare's overconfidence is his downfall; he can't imagine that he could lose. The moral of the story is "slow and steady won the race," which is to say that persistence will overcome talent tainted by arrogant laziness.

Of course, not all plodders are hardworking winners, and not all speedsters are overconfident losers. However, in many cases, such people can't sustain their initial success because they get lazy or burn too many bridges.

Obviously, egomaniacs are not just found among entertainers, although musicians and actors have a well-deserved reputation for behaving like spoiled brats. You would think, though, that we would be a bit more humble in a business where so many careers crash and burn in short order. The music industry is small, especially when you get down to the pros, and it is wise to heed the words of the late Lowell George of Little Feat: "The people you meet on the way up, you might meet up with on the way back down." It pays to be respectful, humble, and cooperative.

Arrogance isn't confined to one's personal talent, though. Even in this age of diversity, one hears musicians put down genres they don't care for. I've heard jazz musicians dismiss country musicians, country and rock players put down "out" jazz, and dedicated improvisers criticize classical musicians who play only written scores.

That's bunk, of course. It takes talent and experience to create music at a high level. You don't have to like or even try to understand all music, but if you don't grasp a genre, you might be best served to withhold judgment about its practitioners as a class.

That brings us to this month's cover story on loop sequencing ("Loop-a-palooza!" on p. 50). Although most loop-sequencing tools are relatively new, the concept of pattern-based composition is ancient, beginning long before tape loops, perhaps in prehistoric drum circles.

Now, loop-based music—a broad term that encompasses numerous musical genres—isn't everyone's cup of tea. I like listening to some of it, and I am familiar with many of the techniques involved. But I admit that I only occasionally listen to loop-based music, and I have no interest in creating it. However, I take issue with those who think that loop-based composition and performance are lesser skills than playing an instrument and composing in more traditional ways. To a loop-meister, a groove machine, software and controllers, or even the entire studio is an instrument. And master loopers can be awesome improvisers. The best loop-based music is among the most vibrant music on today's scene.

So remember the tortoise and the hare. If you're into the looping scene, you're in front of the musical mainstream. Don't be like the hare and let the thrill go to your head. Be patient, hone your skills, and resist the temptation to consider yourself or your work superior because it's cutting edge. If you cleave to more traditional genres, fine. But don't dismiss looping or, for that matter, electronica in general.

You see, music is not a race, and although we should heed the lesson of the fable, we are not tortoises or hares. We each can reach our goals at our own pace as long as we remain dedicated, humble, and hardworking.

ANTHONY PIDGEON

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MAE LAROBIS

MORE OF THE SAME

I just wanted to say something about your article "Demos that Deliver," in the April 2002 issue. That was the first time I ever heard an A&R agent say he or she wants three different songs: a ballad, an up-tempo, and something else.

I've been doing this a while, and I've never heard that before. Most A&R agents, as well as lawyers and managers, that I have spoken to say they want three to four similar songs so that if they like the demo, they know that you have other comparable songs that can be released—no one-hit-wonder-type thing.

So send what you will. But my advice is get a name of an A&R person and then ask if you can send him or her your demo.

Michael Rogers
St. Clair Shores, MI

SCI-FI FICTION

Being a new **EM** subscriber, I just want to confirm that the April issue contains an April Fools' article concerning Dr. Spanky Ganglia's "invention" ("Tech Page: Imagine"). It seems his prior invention was in "Tech Page" in the April 2000 issue.

It presents itself in a convincing-enough manner to be believable at -

first glance, but terribly scary if at all true. The satellite part really was over the top.

Any clarification will be appreciated. If, for some reason, this is serious, where can we get our helmets for shielding the brainwashing from space?

Randy Bundy
via e-mail

Randy—April Fools'!—Steve O

TO THE RESCUE

Thank you for the exceptional article "Panning for Gold" (March 2002). I am a home recordist, and I was hitting a brick wall with the sound of my finished product; your article came to me just in time. It placed my mixing skill at a higher level. Yes, I knew about panning your different instruments to differing degrees left or right, but this gave me in-depth information and a sound base.

Craig H. Hodge
Breddren Production
via e-mail

ALMOST PERFECT

Thanks for another great article by Gary S. Hall ("Burning Ambitions," March 2002). I have only two beefs:

1. He more or less dismissed Multisession for audio use. Although I agree that it's not very useful for audio-only applications, there's an entire article right next door hailing its usefulness for enhanced CDs with first-session audio and second-session multimedia data.

2. He stated, "The CD-R you record is of better quality than any commercial CD." After seven years of testing incoming CD-R masters for replication and commercially pressed discs, I can tell you that that is rarely true. Clean, quality blank media burned at a rea-

sonable speed in DAO mode on a good drive with software-compatible drivers and firmware on a tweaked-out system will probably have lower error rates than some commercially pressed discs. Any other combination will usually fare about the same, if not worse.

Tim Whitlock
via e-mail

CLEAN WINDOWS

In regard to the March 2002 column "Desktop Musician: Windows Washing," I feel compelled to mention a quick and effective technique for freeing up precious resources and avoiding software conflicts. For those people using Windows 98 or better, the System Configuration Utility is a great tool to take advantage of.

To conjure the System Configuration Utility, click on Start and then Run, type "msconfig" in the open line, and then click on OK. If you click on the tab labeled Startup, you will find a list of all applications that fire up every time you boot up your system. Eliminating unwanted TSRs (terminate-stay resident programs) by simply unchecking them will boost your system's resources and reduce the chance of software conflicts. This procedure is a bit less risky than breaking into the registry, as mentioned in the article, because you can simply recheck any listings that have a profound effect on your system's performance.

Other advanced procedures can be performed with the System Configuration Utility, such as removing autoexec.bat and config.sys from the boot sequence, which removes most DOS-compatibility mode drivers and applications that could also cause serious resource conflicts and lockups. Numerous articles online can help guide you through using this utility.

Don't forget to defrag after every major recording session, as there is no

WRH

better way to bog down your system than by running a hard disk full of scattered data. Also, save your work often!

Nez Mirage
via e-mail

A LITTLE HELP HERE

Many thanks to David Battino for his article on producing enhanced CDs, "More than Meets the Ear" (March 2002).

He has inspired me to save a few trees by making my demos enhanced CDs. I'm having difficulty, however. The article states that the preferred method for creating enhanced CDs is to make a multisession CD in which the first session is an audio CD. It also states that, among other software, Ahead Software's Nero is capable of that. Using the latest version of Nero, however, I have been unable to do that. According to Nero, a multisession CD may include only ISO sessions, not audio. Nero's Help file offers no assistance on this point.

If Mr. Battino has created an enhanced CD with Nero, I would appreciate a walk-through. Otherwise, you might want to find out if this is really possible and alert readers before they buy the program with this intention.

Alan Moore
via e-mail

Author David Battino replies: Alan—No audio-CD player of which I'm aware can read a CD with multiple audio *sessions, but the type of enhanced CD I describe in the article has only one audio session, followed by one data session. The audio session can contain multiple* tracks *(that is, songs), which is precisely why the format works: normal CD players see only the first session, but computers see both. To record multiple audio tracks in a single session, your burner and burning software must support* Session-at-Once *mode. Nero does, although it doesn't explicitly call it that. Before burning the audio session, click on the Burn tab, select Disc-at-Once as the write method, and uncheck the Finalize checkbox. Incidentally, Nero can also mount Macintosh SCSI drives, making it a convenient tool for creating hybrid discs.*

LET YOUR VOICE BE HEARD

I enjoy each piece of information you use in **EM.** I am wondering who this Mr. Baker is ("Letters," March 2002), because I know exactly who Mr. Oppenheimer is and how long he has been writing in **EM.** His writing is one of the reasons why people like the magazine and subscribe to it. Please keep saying what you think, in the same way, with the same regularity the rest of us appreciate.

In my country, Venezuela, I have seen how radio stations and many other cultural sources have been "taken" by those corporations—the same as you—and we fight every inch in order to stay active and work for what we believe in.

I always enjoy Mr. Oppenheimer's editorials as well as many other articles by contributors.

Guillermo Carrasco
via e-mail

REVIEWING REVIEWS

I'm a longtime reader of most music and technology magazines that exist, but as time goes on, my interest in these magazines has declined. This is primarily because my major interest lies in the reviews, and most of the information that is covered in reviews is easily available on the Net. (In fact, I've stopped reading SOS completely, because it seems as though the reviews are often literally nothing but a product spec sheet in paragraph form.)

This is frustrating because there's still a great deal of information the expert reviewers have to impart, but if I had a chance to sit down and ask an expert questions about his or her review, I wouldn't be asking about specs—I'd be asking about comparisons to other gear.

I'd like to see a brief paragraph with the salient features, a link to the product home page with the complete specs, a discussion about the sound quality or use of the product, and then a discussion about other similar products and their differences. What I'm really interested in is hearing from someone who owns several pieces of similar gear and can give subjective opinions.

That's just my opinion. Keep up the work on a great magazine!

Eric Hackborn
via e-mail

This changes everything.

kantos 1.0
AUDIO CONTROLLED SYNTHESIZER

From the company that revolutionized vocal intonation processing comes kantos 1.0, a software-based synthesizer that finally liberates you from the tyranny of MIDI, keyboards, controllers or, in fact, anything that stands between you and the music you hear in your mind.

kantos 1.0 is controlled by audio. Any pitched monophonic audio.* Like your voice. Or a musical instrument. Live, in real time. kantos 1.0 analyzes incoming audio and instantaneously extracts pitch, dynamics, harmonic content and formant characteristics. This information is then used to control the kantos 1.0 sound engine. In ways never before possible with a conventional MIDI synth.

To learn more about kantos 1.0, visit our website at www.antarestech.com for audio demos, guided tours and more. Whether you are looking for an alternative to traditional controllers, or are looking to produce electronic music with a level of sonic innovation and dynamic expression that's simply not otherwise possible, kantos 1.0 will, quite literally, change the way you make music.
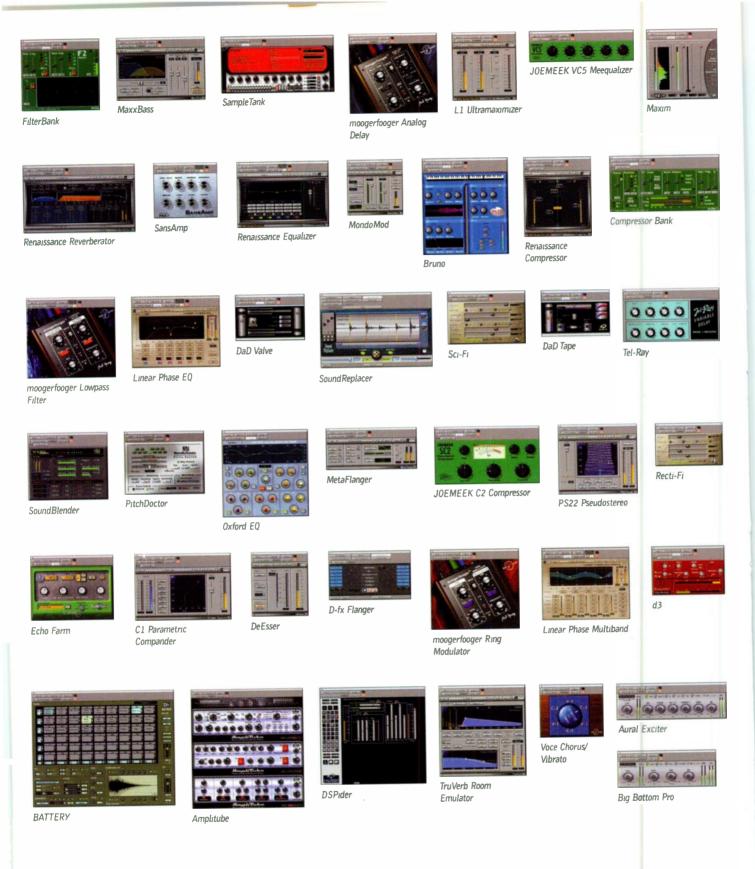
*Actually, kantos 1.0 can also respond to unpitched or polyphonic input. While the output isn't always predictable, it's rarely less than interesting. And particularly with rhythmic input, you can get extremely dynamic (and sometimes downright surprising) results.

KANTOS 1.0 WILL INITIALLY BE AVAILABLE IN MAS, RTAS (MAC) AND VST (MAC) FORMATS, WITH DIRECTX FOLLOWING NOT TOO FAR BEHIND. CHECK OUR WEBSITE FOR DETAILS.

TIMBRAL ARTICULATOR

The heart of kantos 1.0. The articulator takes the harmonic and formant information from the input signal and dynamically applies it to the synthesized signal. It can reproduce the input's characteristics with uncanny accuracy or warp them into a universe of mutant forms.

WAVETABLE OSCILLATORS

Each wavetable oscillator includes its own independent multimode resonant filter and chorus generator. Additional wavetables will be available from our website. Or create your own with pretty much any sample editor.

GATE GENERATOR

The Gate Generator gives you control over the triggering and duration of the envelopes as well as retriggering the oscillators' wavetables. Also included is a variable-threshold noise gate to help clean up any annoying background noise in the control audio.

ANTARES
WHERE THE FUTURE'S STILL WHAT IT USED TO BE

FilterBank

MaxxBass

SampleTank

moogerfooger Analog Delay

L1 Ultramaximizer

JOEMEEK VC5 Meequalizer

Maxim

Renaissance Reverberator

SansAmp

Renaissance Equalizer

MondoMod

Bruno

Renaissance Compressor

Compressor Bank

moogerfooger Lowpass Filter

Linear Phase EQ

DaD Valve

SoundReplacer

Sci-Fi

DaD Tape

Tel-Ray

SoundBlender

PitchDoctor

Oxford EQ

MetaFlanger

JOEMEEK C2 Compressor

PS22 Pseudostereo

Recti-Fi

Echo Farm

C1 Parametric Compander

DeEsser

D-fx Flanger

moogerfooger Ring Modulator

Linear Phase Multiband

d3

BATTERY

Amplitube

DSPider

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Among the many reasons why more and more professionals choose Pro Tools® | HD as their complete audio production environment, here are just a few.

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Microphone Modeler

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Enigma

C4 Multiband Parametric Processor

SynthSpyder

UltraPitch

Audiotrack

PAZ Psychoacoustic Analyzer

Scratch

Speed

Pultec EQP-1A

d2

Analog Channel

SuperTap

moogerfooger 12-Stage Phaser

Q10 Paragraphic Equalizer

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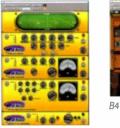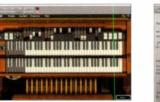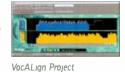
For the most up-to-date information on Digidesign Development Partner products for the Pro Tools|HD platform, visit www.digidevelopers.com. For information on the new Pro Tools|HD systems, visit www.digidesign.com or call 1.800.333.2137.

digidesign

WHAT'S NEW

By Marty Cutler

RØDE NT4 AND NT5

Røde's NT4 ($899) is a stereo condenser microphone with a pair of ½-inch cardioid capsules in a 90-degree XY pattern. The mic comes with XLR and minijack connectors, and it runs on 48V phantom power or a 9V battery.

The Røde NT5 ($599) is a cardioid condenser microphone designed for instrument recording. The manufacturer touts its versatility, claiming suitability for guitar, flute, violin, reeds, and drums. The NT5 requires 48V phantom power, and it can be ordered in matched pairs with a custom carrying case, two standmounts, and two windshields.

The manufacturer reports a frequency response of 20 Hz to 20 kHz and a dynamic range of >128 dB for both microphones. The maximum SPL is 143 dB at 1 percent total harmonic distortion. Røde Microphones; tel. (310) 328-7456; Web www.rodemicrophones.com.

▶ TERRATEC MIDI MASTER

The MIDI Master ($399) from TerraTec provides 49 Velocity-sensitive keys and a USB port for connecting directly to your computer. The keyboard also has a MIDI Out, so it can control MIDI devices without a computer.

In addition to a Data Entry slider, the MIDI Master has both a modulation wheel and a pitch-bend wheel. The Data Entry slider can be assigned to MIDI Control Change messages. You can select any of 13 Velocity curves, and you get buttons for transposing MIDI output and a switch for sending System Reset messages.

The keyboard features a ten-key pad for programming and ten nonvolatile user-memory locations for storing your control assignments. The MIDI Master can draw power from its USB port. However, you can use an optional power supply for standalone performance. TerraTec bundles USB drivers for Windows 98, SE, ME, 2000, and XP, as well as drivers for Mac OS 9. Fostex Corp. of America (distributor); tel. (562) 921-1112; Web www.terratec.net.

▼ STEINBERG CUBASE SX

Cubase SX (Mac/Win; $799.99), from Steinberg, sports a redesigned audio engine and user interface, graphical parameter automation, surround-sound support, and more. The audio engine is based on code from Steinberg's Nuendo, and it can handle six channels of surround sound (stereo, "quadro," LRCS, 3/2, and 5.1).

Cubase SX fully supports Steinberg's VST System Link, which lets you network computers to share heavy processor loads imposed by host-based digital-audio processing. For example, synchronization, transport, and MIDI data can be managed from one computer.

Cubase SX's Project Window replaces the Arrange Page and offers real-time drag-and-drop placement and editing of audio and MIDI data. In addition, you can have multiple projects open simultaneously. The Track Inspector now lets you add insert effects and adjust aux sends, pan, and volume without opening the mixer. A new toolbar provides access to commonly used editing tools, shortcuts to editing windows, and a built-in transport bar.

The redesigned mixer section offers three views: Standard, Small, and Extended. Individual widths can be defined for each channel view. It is now possible to bounce tracks to disk in real time or offline, which lets you use more effects than your host system would allow in real time.

The sequencer is compatible with earlier versions of Cubase, so you can import Cubase-created Songs, Arrangements, or Parts. Cubase SX supports MIDI plug-ins and includes an arpeggiator, a MIDI compressor, an echo processor, and a microtuning plug-in. You can also use Cakewalk MIDI plug-ins with Cubase SX.

Cubase SX requires a Pentium III/500 MHz computer with 256 MB of free RAM and Windows 2000 or XP, or a Macintosh PPC G3/350 MHz machine with 256 MB of free RAM and Mac OS X. Steinberg North America; tel. (818) 678-5100; e-mail info@steinberg.net; Web www.us.steinberg.net or www.cubase.net.

A new perspective on [*the*] bottom end.

Introducing *Laminar Spiral Enclosure*™ Technology:
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The Truth - in black and white: a revolutionary, proprietary technology [*and shape*] that will forever change your expectations of what an accurate, multi-channel professional surround-sound monitoring environment should sound like. The new Genelec **7000 Series** consists of four (4) active subwoofers — painstakingly engineered to enhance the entire range of our active monitoring systems — from the award-winning **1029**'s right up to the formidable **1039**'s.

These new subwoofers look different for a very good reason: *Laminar Spiral Enclosure* technology. **LSE**™ is evident in the unique circular shape of the enclosures, identical in execution for the 8", 10", 12" & dual-driver 12" models. This radical design departure dramatically alters how high-level, low-frequency acoustic energy is delivered to virtually any critical-listening acoustic space.

The main benefit? The compromises inherent in traditional ported enclosures are now things of the past; the unrestricted *laminar flow* of low-frequency energy emanating from the rigid, tuned enclosures of the **7000 Series** is truly an extraordinary listening experience. When combined with the latest 6.1 electronic bass-management technology, Genelec *Laminar Spiral Enclosures* not only turn things around, they give the professional a whole new perspective on the meaning of superior performance.

Model 7070A Active Multichannel Subwoofer – 12" dual voicecoil driver, 19Hz to 120Hz, 114dB with 6.1 Bass Management feature set. There are two smaller models (7050A & 7060A) and one larger system (7071A).

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Four Reasons Why Fastest and

As the first software solution to use streaming technology for playing huge samples off your PC's hard drive, **TASCAM GigaStudio**™ quickly became the world's leading sampler for professional composers and musicians. But today, there are a few more choices in samplers that stream audio. Why is Giga still considered the best sampling tool for power users? Here are four reasons to keep in mind:

1. Kernel-Level Functionality. Giga offers something that no plug-in sampler can provide: it operates at the kernel level of your PC, beneath the point where Windows® functions. This patented technology allows GigaStudio to offer the very best latency specs, even with huge amounts of polyphony.

2. Sample Buffering. No hard drive on the planet is fast enough to stream samples with the blazing speed needed for real-time performance. Giga pre-loads a tiny portion of your samples into RAM to accomplish this task. Plug-in samplers have to load a much larger section of the sample into RAM, which can greatly affect the number of instruments that can be used in one session.

3. Standalone Application. GigaStudio can be used on a PC by itself as a standalone software application, or can be used on the same machine as your PC-based sequencer. Plug-in samplers need to be accessed within a DAW application, which can severely limit their professional performance capabilities.

4. The Best Libraries = The Best Sound. Put all the technology aside for a moment, and all that really matters is the sound. Giga libraries were specifically designed for the Giga platform. They take advantage of Giga's Dimensions functions like key switching, crossfading, velocity control and more. Plus, their incredible detail and sonic quality is optimized only in GigaStudio...with nothing lost in the translation.

So if you're into the very best that sampling technology has to offer, get into Giga. Visit your TASCAM dealer or check it out online at **www.tascam.com**, because when it comes to sampling, bigger and faster is always better. Period.

Giga Sound Library Spotlight

TASCAM/Christian Lane Ultimate Marimba & Vibes
NEW! With four sets of mallets to choose from, this 2-disc library features a great-sounding Marimba One 5-octave instrument and a Musser set of vibes. Multiple velocities, rolls, muted strikes and lots of extras truly make this the ultimate!

TASCAM/Jim Corrigan Nashville High-Strung Guitars
One of the coolest, most playable acoustic guitar collections for Giga. Recorded with incredible quality, this totally authentic collection of up and down strums and dynamically playable single strings for solo parts represents the sound of Nashville at its finest.

TASCAM/Christian Lane Ultimate Timpani
NEW! A 6 CD library built from a set of Hinger Touch Tone timpani with calf heads. Five mallets, eight velocity levels, individual right and left hand samples and incredible detail give this library a 5/5 star rating in Electronic Musician magazine!

TASCAM/Peter Ewers Symphonic Organ
The entire, historic, grand La Madeleine, Paris cathedral and the Cavaillé-Coll organ for GigaStudio! For the first time ever in any sampled pipe organ, the original cathedral ambiance is included via release triggered samples.

TASCAM/Conexant GM150/GM500 General MIDI Kits
You've never heard General MIDI like this! Two different collections (150MB and 500MB) of multi-megabyte instruments, including a complete set of acoustic instruments and synthesizer textures.

TASCAM/Gary Garritan GigaHarp
The sound of angels! Every string of a Salvi Pedal Harp sampled in stereo representing seven pedals, four attacks/velocities per string, two harmonics per string, glissando, hand-dampening and muffling.

SOUND ADVICE ▲ ▲ ▲ ▲ ▲

▲ Q UP ARTS

If you have ever wanted to use the drum tracks from *Cheap Trick at Budokan,* here's your chance. In *Bun E. in a Box* (audio CD, $99; WAV, Akai, Giga, or AIFF, $199; 24-bit WAV or Halion, $249), drummer Bun E. Carlos provides multiple loops for every Cheap Trick hit, as well as individual drum samples from his own collection of instruments. Renowned engineer Steve Albini recorded and produced the sampling session.

Voices of the Aztecs (audio CD, $99; WAV, $249; Akai, SampleCell, or Halion, $299) is more than a vocal collection; you also receive a compilation of Aztec instruments and loops. The 16-bit, 44.1 kHz instrument samples were recorded in Mexico and come from artifacts found in archaeological digs. Among the instruments included are the *huehueti* drum, ocarinas, water drums, wood-percussion instruments, clay flutes, and a seashell horn. Q Up Arts; tel. (800) 454-4563 or (801) 486-8225; e-mail info@quparts.com; Web www.quparts.com.

▶ FORWARDINOUTBACK

A panoply of fresh didgeridoo and percussion samples and loops is available on *Percussion Samples Volume 1, Didgeridoo Samples Volume 1.2* (WAV; $66). The two CD-ROMs also offer kits arranged for Native Instru-

ments Battery and banks for the MAZ Virtual Sampler. All samples are 44.1 kHz, 16-bit files.

There are samples and loops using *udu* (clay pot), *kalimba, gopichand,* bodhran, frame drum, jaw harp, singing bowl, shakers, and assorted grooves. All instruments except the *gopichand* were played by Ralph "Stick" Hermann.

The CD-ROM also provides a Sample Player application for previewing the samples, a PDF booklet with complete documentation, and a Read Me file that contains application suggestions and technical information. Forwardinoutback; tel. 49-30-455-8432; e-mail info@forwardinoutback.com; Web www.didgeridoo-samples.com.

▶ BIG FISH AUDIO

Live drumming is featured on the three-CD collection *L.A. Riot: The Funky Drummers* (audio CD; $99.95). *Funky Drummers* serves up 207 tracks of full song arrangements that are broken down by intro, verse, chorus, breakdown, and chorus out.

Chronic Music, creator of previous *L.A. Riot* collections, produced the set, engaging the talents of four studio drummers. Some of the tracks present two drummers playing simultaneously for grooves that would be impossible to create with a single player. Big Fish Audio; tel. (800) 717-FISH or (818) 768-6115; e-mail info@bigfishaudio.com; Web www.bigfishaudio.com.

▼ SOUNDFOREST

The two-CD set *S2 Sonic Scenes . . . from the Edge* (AIFF/WAV; $59.95) offers 860 MB of samples that lend themselves to film scoring. Samples are as large as 20 MB in size, and all clips either contain loop markers or work as complete loops. Producer Michael Kelsey characterizes much of the collection as textural, and sometimes bizarre, in nature.

The sounds are organized into folder categories, such as Building Tension, Cat and Mouse, and Environment of Transformation, that suggest cinematic elements. Soundforest includes a Microsoft Access/Excel-compatible database with sortable fields, including Category, Play Style, Loop Placement, Loop Start and End Points, and Name. Soundforest; tel. (877) 237-5771; e-mail info@soundforest.com; Web www.soundforest.com.

From mic-pre to CD...

DPS 24 DIGITAL PERSONAL STUDIO

The DPS24 is the only affordable integrated hardware digital workstation that offers 24 tracks of recording without data compression. Most types of data compression throw out portions of your audio during recording, and use a mathematical algorithm to approximate the original audio upon playback.

Data compression can adversely affect your audio quality and stereo imaging, especially with multiple generations of track bouncing.

Kind of sounds like your old cassette multi-track, doesn't it?

It's important to know what you're getting when you invest in any recording solution. The DPS24 was designed from its inception as a professional production tool and not simply a scaled-up portable studio.

We combined a 24bit/96kHz linear 24-track hard-disk recorder, a 46-input moving-fader automated digital mixer, a sample-accurate multi-track graphic waveform editor, 4 stereo multi-effects processors, ak.Sys TrackView and VST plug-in platform software, and a CD mastering and archiving suite, without creating the performance or user interface compromises found in many integrated workstations.

Utilizing our Q-Link navigation design, the DPS24 offers access to any major function with one button press. The Q-Channel strip of automated LED rotary controls enables instant access to any channel strip on the mixer.

Features like two banks of inputs to eliminate re-patching, balanced channel inserts which enable external mic preamps to bypass the on-board preamps, multi-function Q-Knobs for realtime effects control, and up to 24 channels of ADAT I/O offer professional production capabilities that give you the real-world advantages you need to bring your artistic vision to its full potential.

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▼ APOGEE MINI-ME

Apogee's Mini-Me ($1,395) combines a 2-channel mic preamp, two channels of 24-bit, 96 kHz A/D, and a line-level input in one portable device. The Mini-Me lets you choose sampling rates of 44.1, 48, 88.2, or 96 kHz. When it's

connected to your computer's USB port, the Mini-Me automatically converts incoming data to the selected sampling rate. You can also choose 16- or 20-bit output at sampling rates of 44.1 or 48 kHz. The Mini-Me uses Apogee's UV22 HR dithering with word-length reduction.

The front panel has independent left and right level controls as well as a knob that controls the monitoring level. A three-way front-panel switch allows you to activate the unit's built-in compressor-limiter and to select between Soft-Limit/Compress and Soft Limit modes. A second switch lets you select three preset compressor/limiter curves. The Mini-Me provides a knob for choosing sampling rates and bit depth.

The unit also has a 48V phantom-power switch and a button for selecting stereo or mono output. In addition, the button serves as a knob for balancing the direct-input audio with audio returning from the USB port.

Inputs are on balanced XLR/TRS combo connectors; the Mini-Me automatically switches to line-level input when you insert a TRS plug. Other connectors include AES/EBU and S/PDIF outputs, and an ⅛-inch stereo headphone jack for monitoring. The Mini-Me can draw power either from the USB port or from the provided external power supply. Apogee Electronics Corp.; tel. (310) 915-1000; e-mail info@apogeedigital.com; Web www.apogeedigital.com.

▶ ANTARES KANTOS 1.0

Any monophonic audio signal is fodder for Kantos 1.0 (Mac/Win; $395), a software synthesizer plug-in for MOTU Audio System, VST, and Real Time Audio-Suite from Antares Audio Technologies. (A DirectX version should be available by the time you read this.) Interestingly, the synth doesn't use MIDI control, but rather pitch, dynamics, and timbre information from an external audio source. Default settings route pitch information to the oscillators, dynamics control the amplifier, and harmonic content controls the Timbral Articulator. The software, however, features a sophisticated modulation matrix, letting you route audio signal components to other parameters.

Kantos is loaded as an insert effect on your digital audio sequencer. You get an input-level control with measurements in decibels, as well as a real-time display of the signal's amplitude. A Gate Generator allows you to adjust threshold levels and sustain time for the signal. You also get an adjustable noise gate to remove any background signal noise.

Kantos offers a pair of wavetable oscillators and comes bundled with a collection of wavetables; you can create your own wavetables with just about any digital-audio editing software that supports AIFF file formats. Furthermore, you can download new wavetables from the Antares Web site. Kantos also includes a

noise generator and a pair of sine-wave generators that track the wavetable's fundamental pitch. A Pitch Constrain keyboard lets you limit the pitch output of the signal. For example, you can constrain the output of the melody to a diatonic scale and remove vibrato from the signal.

The Timbral Articulator analyzes the harmonic content and formants of the audio input; you can use the Formant Offset and Resonance controls to reshape the audio signal's tonal characteristics as well as those of the wavetable. You also get a multimode resonant filter for each oscillator.

It's designed to process monophonic audio tracks, but you can feed polyphonic tracks into Kantos with interesting and useful—though less predictable—results. Even though audio input can control sound dynamics, Kantos gives you two ADSR envelope generators (EGs): a defeatable amplitude EG and an assignable modulation envelope. Envelope values are displayed in milliseconds. The synth also provides a pair of multiple-waveform LFOs, with frequencies indicated in hertz.

Kantos offers a variable-feedback de-

lay line, which allows you to create rhythmic effects. You can set the delay tempo in bpm or with tap tempo. The output section features a submixer, which controls the internal sound sources, and a main mixer, where you can balance processed and unprocessed signal. You can load Kantos as a mono-in/stereo-out plug-in. You can load as many instances of Kantos as your processor can handle to

create more straightforward polyphonic material.

Kantos requires a host program that supports at least one of the plug-in formats mentioned previously. Antares Audio Technologies; tel. (831) 461-7800; e-mail info@antarestech.com; Web www. antarestech.com.

REV UP ▲ ▲ ▲ ▲ ▲

▲ SONIC FOUNDRY

Among the new features in Sound Forge 6.0 (Win, $499.95; upgrade from version 5.0, $129.95) are sample-accurate editing and background rendering that lets you edit one file while the program processes another.

The DirectX Plug-In Manager allows you to easily sort, edit, and arrange plug-ins. Sample-accurate editing is supported with the enhanced Time Zoom feature and the ability to snap to a specific sample.

Version 6.0 features new video-rendering capabilities, including high-quality video resizing, a Stretch to Fill Frame command, and the ability to resample source video. Sound Forge now has full support for files over 4 GB (if your computer supports the NTFS file system). Sonic Foundry bundles Vegas Video LE 3.0, which offers eight tracks of video, effects and transitions, and the ability to simultaneously time-stretch audio and video. Sound Forge 6.0 requires a Pentium II/ 200 MHz, 32 MB of RAM, and Windows 98SE, ME, 2000, or XP. Sonic Foundry; tel. (800) 577-6642 or (608) 256-3133; e-mail customerservice@sonicfoundry .com; Web www.sonicfoundry.com.

▶ ARTURIA

Storm 2.0 (Mac/Win, $199; upgrade from version 1.0, free) adds a host of new features to this popular program. The Hall lets users collaborate online. The initial version of this fea-ture allows peer-to-peer file sharing and offers a chat room and an assortment of tips and tutorials.

The program's Composition Wizard contains song templates for different musical styles. Storm 2.0 adds a vocoder, a compressor, and the Shadow module, an eight-note polyphonic, analog-modeling VST synth with two oscillators, pulse-width modulation, and a multimode filter.

Some of Storm's synthesis and effects algorithms are redesigned. Arturia has improved its sample libraries, and a bug in the sample-loading process has been repaired. Synthesis modules offer graphically consistent interfaces and redesigned onscreen ergonomics. PC users

need a Pentium II/300 MHz computer, 64 MB of RAM, and Windows 98, 2000, ME, or XP. Mac users need a PPC G3/300 MHz, 64 MB of RAM, and OS 9 or OS X. Arturia; tel. 33-438-020-5 55; e-mail info@ arturia.com; Web www.arturia.com.

▶ CAKEWALK

Cakewalk has improved the editing features and user-interface design and added support for ReWire 2.0 in Sonar 2.0 (Win, $479; upgrade from version 1.0, $79) and Sonar XL 2.0 (Win, $599; upgrade from version 1.0, $169). The upgrade includes Cyclone DXi, a groove-oriented sampling plug-in. Cyclone breaks sampled grooves into indi-vidual slices and lets you reorganize the slices or impose rhythmic feels from one groove onto another. You can trigger 16 grooves with external MIDI gear, a sequenced MIDI track, or a mouse. Cyclone DXi supports multiple audio outputs.

You can run any ReWire-compatible device within Sonar, choosing instrument channels by device. You can add Cyclone loops or other audio tracks to Steinberg's Reason, for example.

Sonar 2.0's new Graphical Drum Editor offers a drag-and-drop grid with editable Velocity tails for each drum note. The editor supports composite drum kits from multiple MIDI devices.

Sonar 2.0 provides dedicated support for the CM MotorMix and Tascam's US-428 and US-224 control surfaces. If you use another control surface, the program's Global Controller Module offers presets for other units and a Learn mode to accommodate unspecified controllers.

Sonar 2.0's Synth Rack provides improved integration of DXi software synths. You can mute or solo groups of synths. A macro provides single-step synth-track inserts, allowing you to patch soft synths into track inputs.

Sonar 2.0 has third-party sound libraries, DXi soft synths, and DirectX 8 automated effects. Requirements are a Pentium III/500 MHz computer, 64 MB of RAM, and Windows 98SE, ME, 2000, or XP. Cakewalk; tel. (888) CAKEWALK or (617) 423-9004; e-mail sales@cakewalk .com; Web www.cakewalk.com.

▶ ANALOGUE SOLUTIONS VOSTOK

The Vostok ($2,595), from Analogue Solutions, resembles the coveted EMS VCS3 synthesizer, with its suitcase-style design and front-panel patching matrix. However, the similarities end there.

The Vostok offers a built-in MIDI-to-CV converter, two analog oscillators, one digital wavetable oscillator, and more.

The patching matrix gives you 484 patch points. Patching is localized in a small area on the front panel, enabling you to create complex patches without spreading patch cords all over the face of the unit. The matrix also provides patch points for connecting external devices.

The Vostok features a variety of control modules. You get two control-voltage (CV) modules: CV1 is hardwired to pitch and CV2 can be assigned to different functions, including gate, accent, and legato. Vostok sports a two-way joystick controller and a built-in, eight-step analog sequencer.

The Vostok has two voltage-controlled oscillators that can generate pulse, sawtooth, and square waves. You can modulate pulse width, and both oscillators offer sync and glide. The third oscillator has a digital wavetable with 256 waveforms. A '70s retro-style LED shows the selected wave sample number and the MIDI Note Number when played. A white-noise generator is also included.

The synth offers two multimode voltage-controlled filters (VCFs) that are based on the circuitry of the Korg MS20. The filters offer highpass and lowpass configurations with independent frequency cutoff settings, and you can combine the two filters to get a bandpass filter. There are two ADSR envelope generators; a dial lets you adjust envelope level continuously from full-on to inverted. The two LFOs offer a variety of waveforms, as well as voltage control over frequency. LFO 2 adds sample and hold, with and without a slew control.

The Vostok's amplifier module is wired in series with the VCF and offers ring and amplitude modulation. Output is on an unbalanced ¼-inch output jack. The suitcase enclosure has key lock latches with a carrying handle and a lift-off cover. Enport (distributor); tel. (402) 398-0198; e-mail info@analoguesolutions.com; Web www.analoguesolutions.com.

▶ VIRSYN TERA

Tera (Mac/Win; $279) gathers 16 independent modular synthesizers, a sequencer, and a mixing console with an effects rack into a single software package. The synthesizer works as a VST2 plug-in or as a standalone instrument.

The 16 independent synths can employ one of five types of synthesis: virtual analog, waveshaping, physical modeling, FM, or spectral synthesis. Spectral synthesis deploys 128 resonant filters with real-time dynamic controls.

The Effects Rack offers as many as 48 effects at once. Effects include chorus,

flange, phaser, distortion, echo, delay, and reverb. Effects appear as inserts in the mixing console.

The sequencer assembles songs from patterns, and patterns are created with the step sequencer or live input. You get as many as 64 steps per sequence. There is also a programmable arpeggiator, and you can control and dynamically automate as many as 64 parameters.

Windows users need a Pentium II/400 MHz computer with 64 MB of RAM; Windows 98, ME, 2000, or XP; and a DirectX and ASIO-compliant sound card. For the Mac, you need a PPC G3/300 MHz machine with 64 MB of RAM and Mac OS 8. By the time this goes to press, a Mac OS X version should be available. VirSyn Software Synthesizer; tel. 49-7240-202-956; e-mail info@virsyn.com; Web www.virsyn.com.

▶ STEINBERG VIRTUAL GUITARIST

The Virtual Guitarist ($249) is a sample-playback VST Instrument dedicated to rhythm-guitar parts. The instrument comes on three CDs, providing more than 2 GB of styles, sounds, and playing techniques.

The guitar techniques include fingerpicking, folk-style strumming, wah-wah rhythm parts, and heavy-metal riffs. The guitars are divided among two VST Instruments (electric and acoustic) and include Spanish, Steel String, Clean Strat, Wah, and Metal.

Steinberg drafted professional guitarists to lay down the styles and techniques; however, the virtual bodies of the instruments are adorned with guitarlike knobs that let you adjust performance parameters. The Virtual Guitarist allows you to vary phrasing, instrument tone, swing, and dynamics in real time. The program is compatible with Cubase, Nuendo, and other VST-compatible host programs.

Steinberg North America; tel. (818) 678-5100; e-mail info@steinberg.net; Web www.us.steinberg.net or www.cubase.net.

WEB PAGE

By Gino Robair

WEB SITE OF THE MONTH

Looking for a way to get past a pesky problem? Trying to get your creative juices flowing? Let Brian Eno and Peter Schmidt help.

In 1975 Eno and Schmidt created *Oblique Strategies*, which is a deck of cards that contains aphorisms to help artists get through the rough patches of creativity. Subtitled "Over one hundred worthwhile dilemmas," the collection of statements and questions ranges from the obvious ("Abandon normal instruments" and "Use filters") to the obscure ("It is quite possible [after all]" and "Do nothing for as long as possible").

Over the past 27 years, the deck has been immensely popular and has gone through four limited-edition printings. These command a pretty penny on the auction market, and until recently, it has been difficult for the starving artist to get hold of the cards. Enter the Internet.

Begin with a visit to EnoWeb (http://music.hyperreal.org/artists/brian_eno) for an overview of Eno's world, as well as a brief introduction to *Oblique Strategies*. For more in-depth coverage, visit Gregory Taylor's Oblique Strategies site (www.rtqe.net/ObliqueStrategies), which contains a history of the cards, an online version, and links to related applications.

These include a charming Mac OS hyperstack version; an Apple OS applet version (click on the desktop icon and a single strategy comes up); archived versions for Windows 95 users; and even one for Palm Pilot designed by Lonnie Foster, which allows you to either choose which edition of the deck you use or make your own deck.

Other versions include a JavaScript version (www.topix.com/~sean/oblique.htm), a simple but elegant Java version (www.asahi-net.or.jp/~rf6t-tyfk/oblique.html); and a fancier Shockwave version (http://members.telocity.com/~bryrock2/eno/oblique.html), which can be viewed from the Web or downloaded as a desktop version for Mac and Windows NT.

For readers who are interested in obtaining the treeware copy of *Oblique Strategies,* a fifth edition of the cards is now available. Check EnoWeb for information on how to order.

DOTDOTDOT.COM

If you're interested in getting your hands dirty, check out the **E-Music DIY Archive** (www.physicsenterprises.andrews.edu/diy_archive). The site includes schematics for amplifiers, filters, and oscillators; manuals for vintage gear from ARP, Korg, Moog, and Oberheim and for a few popular music-related chips; tips about soldering and desoldering; reference guides to capacitors and operational transconductance amplifiers; and a tutorial on customizing resistors. Even if you haven't cultivated the geek within you, Music DIY archive is worth a visit. . . . **Broadjam** (www.broadjam.com) now offers a music database and search engine. The Song Management System allows you to search for songs based on mood, genre, era, language, key, hook (the most memorable part of the song), and similarity to other artists. The search results include a digital file that provides streaming audio to the user. . . . **Softsynth.com** (www.softsynth.com) has just released JavaSonics (www.javasonics.com), which gives 16-bit audio I/O capabilities to Java applets and applications. The Mac/Win-compatible JavaSonics plug-in is free, and the JavaSonics SDK is free for personal, noncommercial use. . . . Are you interested in getting music-related news in a timely fashion? Do you want to be the first on your block to hear about new products as they're announced? Then check out the Electronic Musician online newsletter, **EMusician Xtra** (www.emusician.com). Our biweekly newsletter contains important music news from Reuters, information about new products
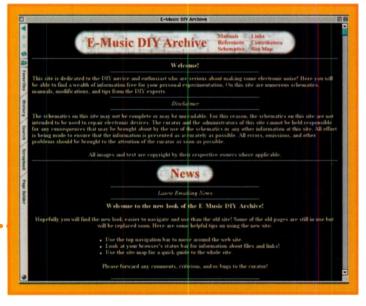
and software updates, links to interesting Web sites, and more. Recent news items include updates on Napster's appeals, information about using the Apple iPod with Windows, and an obituary for electronic-music pioneer Oskar Sala, as well as reports about new equipment from the NAMM and Musikmesse trade shows. To subscribe to EMusician Xtra, visit **EM**'s Web site.

DOWNLOAD OF THE MONTH

VST Shell and MultiVST Shell from Wildfrontear (www.wildfrontear.co.uk) allow you to use VST instruments as standalone applications or in MIDI software that does not support VST (such as Digidesign's Pro Tools). VST Shell supports a single VST plug-in instrument, and MultiVST Shell (shown here) supports four instruments and four VST effects plug-ins. Both programs were written in Max/MSP and require the Max/MSP run-time environment (included with the download) and Open Music System (OMS) 2.3.8 (widely available on the Web) to operate. Minimum requirements vary with the plug-ins being used, but a Macintosh G3/233 MHz or higher with at least 128 MB of RAM and OS 8.6 or above are recommended.

MultiVST Shell offers a number of features beyond simple VST support. They include an onscreen keyboard that will operate in Mono or Poly mode, a basic Standard MIDI File player, and provisions for recording the audio output to disk. Each of its four VST plug-in channels can have its own MIDI channel assignment and can feed a separate output if your

sound card and drivers support that. There are also panel controls for MIDI Program Changes and a user-assignable MIDI continuous controller. Additionally, all VST instrument and effects control panels can be opened for editing.

Wildfrontear provides a number of free, downloadable add-ons including several VST plug-ins, an arpeggiator, a 16-step sequencer (Seq16), and a six-voice drum machine (Drum16) with built-in sample player and synth. These can be used as standalone applications to play other OMS-friendly applications or combined with MultiVST Shell as a virtual VST band. Whether you just want to noodle on your favorite VST synth or get your favorite non-VST sequencer into the VST action, Wildfrontear is worth a visit.

—*Len Sasso*

WEB APP

Although MP3 is the most popular format for distributing Web audio, one drawback is that a listener's Web browser will open a new browser window with a third-party player when an MP3 file is selected. Not only is the new window confusing to some users, it often takes away from the presentation of your Web page.

Rather than leave audio playback to the mercy of the browser, Web-page designers can embed the Macromedia Flash player in a Web page so that sounds play without a window being launched. By using the Macromedia Flash SWF format for your audio files, you can choose during the conversion phase whether your music streams or downloads. However, only the Flash authoring application can create both streaming and downloadable SWF files; the third-party applications, below, create only streaming files.

Converting MP3 files to SWF files without using the Flash software has some advantages. You can achieve higher-quality sound in a Flash movie by creating an MP3 using a professional audio editor, such as Sonic Foundry Sound Forge or BIAS Peak,

instead of the lower quality Flash MP3 encoder, before converting your MP3 to SWF using a third-party application. Also, third-party apps let you batch-process MP3 files to SWF and preserve the copyright information (ID3 tags) from the MP3 file in the Flash presentation. The Flash authoring software does not support batch processing or the transferring of ID3 tags from MP3s.

Ueberschall Sample Service's SoundNails (Win; www.soundnails .com) is shareware that converts MP3 files into the SWF format. It's great for musicians who want a quick and easy way to publish songs online and take advantage of the large audience for streaming-audio content. Sound-Nails creates Web-ready HTML code, does batch conversion, and generates a Web page with embedded Flash players and ID3 tags for each converted song. You can control the look of the page by choosing from several player-console designs and HTML templates.

The SoundNails Easy version ($10) can batch-process as many as five MP3s and comes with five player consoles and page layouts. The Pro version ($65) can batch-process more than five MP3s and supports customizable player consoles and HTML templates. SoundNails runs under Windows 95, 98, and 2000.

Grooveware Multimedia's SwiffPEG (Win; www.swifftools .com) lets you merge an MP3 file with an existing SWF or export MP3s as a standalone sound-only SWF file that requires no player console. That's useful for musicians who need to deliver soundtracks to Flash developers in the SWF format. The free version of SwiffPEG can batch-process and merge MP3 files while keeping ID3 tags intact in the output SWF. With the registered version ($25), you can also create sound-only SWF files. SwiffPEG runs under Windows 95, 98, and 2000.

—Hayden Porter

BAND ON THE WEB

Although the terms *avant world* and *ambient* have been used to describe Straylight (www.straylight.ws), the band steers clear of world-beat clichés. Instead, Straylight—Charles Cohen (Buchla Music Easel), Jason Finkelman (berimbau, riti, percussion), and Geoff Gersh (guitar)—focuses on free improvisation, exploiting the timbres of their instruments while maintaining a rich ensemble sound.

"Straylight is dedicated to the art of improvised music," explains Finkleman. "The exploration happens primarily during live performance, and our process of composing in the moment is rooted in listening carefully to one another."

The combination of electric guitar and synthesizer with acoustic percussion and single-stringed bowed instruments gives the group an appealing rawness that is very fresh. Straylight's eponymous first release, on the Deep Listening label (www .deeplistening.org), offers two studio tracks and a live set from the Knitting Factory in New York.

Gersh's guitar playing runs the gamut of styles, from loop-based drones to fuzz-soaked leads. For example, on the studio piece "Riti Driff," Gersh creates a rhythmic fabric of overlapping guitar parts while Finkelman works out on the riti. Behind him, Cohen adds elaborate and well-placed filigree on the Buchla.

Although improvising with a synth is nothing new, doing so with a modular analog synthesizer is unusual these days and not for the faint of heart. Cohen's instrument, Buchla's suitcase model featuring 200-series modules, includes a complex oscillator, a sequencer, a pressure-sensitive keyboard, and customizable program cards for retrieving patches. However strange the synth's interface is by today's standards, Cohen has complete mastery over it. And he's probably the only musician actively using the instrument today.

"I've been playing the Buchla Music Easel for 26 years," says Cohen. "With its color-coded slide pots, its musically logical panel layout, and its sculptural patching system, I can comprehend the state of the instrument with a fleeting glance. In improvisation, I try to think less and experience more. The primary experience for me is to listen. Next, I want to respond quickly, honestly, playfully, and intuitively. The touch-sensitive, capacitance-activated keyboard responds smoothly to the slightest skin contact, and its assorted CV outputs can be directed to sonic and structural parameters very quickly. The potential to flow supplely into and amongst all the basic electronic sound forms is literally at my fingertips." ●

PHOTO COLLAGE BY MIKE SPERANZA

By Matt Gallagher

One for Al

Al Schnier's solo debut emerges from spare moments on the road.

Guitarist-songwriter Al Schnier grew up absorbing an abundance of musical influences while learning to play a number of instruments. Since 1991 he has largely plied his trade with moe, an improvisational rock group and one of the hardest-working bands on the jam-band scene. Moe spends much of its time on the road fearlessly exploring and fusing American musical styles—hard rock, prog rock, blues, folk, country, bluegrass, and funk, to name a few.

In recent years, Schnier acquired an interest in synth music that culminated in his first solo release on moe's Fatboy Records, *One,* an instrumental album of chilled-out electronica and dance music. "A lot of music that I had been listening to—stuff like Stereolab, Tortoise, the Chemical Brothers, Moby, things like that—inspired me to go ahead and do this," Schnier says. "It's not enough for me just to listen to the music; I want to know how it works."

Schnier assembled a mobile studio to bring on tour with moe in the summer of 2001 so that he could sketch out song ideas. He brought his Mac G3/400 MHz Lombard PowerBook, equipped with a FireWire card and loaded with Propellerhead Software's Reason and Steinberg's Cubase VST/32. He also packed a RolandED PC-300 USB MIDI keyboard controller and a Mark of the Unicorn 828 computer audio interface.

"I had this very portable setup," Schnier says. "I was working on the bus, backstage, and in hotel rooms. Before the summer was over, I was four or five songs into the album already, and I didn't even plan on starting it until January when I [was planning to] set up a desktop system [at home], so I just went with it at that point.
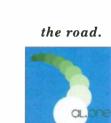
"I did most of the composing, arranging, and recording on the road," he continues, adding that Reason was his primary songwriting tool. "The arrangement process is very user-friendly because you can cut and paste sections and add parts by getting some of your existing machines in the program to play other machines—you can get the drum machine to play synths and vice versa. You can patch anything anywhere in this program."
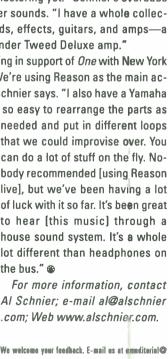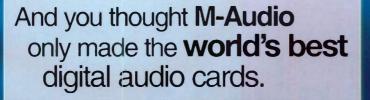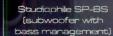
Schnier also recorded audio sources on the road. "I had to dump everything into Cubase and do all my audio there," he says. "If I had time backstage, I could work out some guitar parts or some of the real analog synth stuff that I wanted to add. I have a Minimoog and a [Roland] Juno-60 that I use with moe. Once I moved everything over to Cubase, I'd stay in Cubase because that's where I mixed the album.

"At home I did some final overdubs, editing, and mixing, and then I sent it off to be mastered; I wasn't ready to tackle mastering yet." Schnier's overdubs introduced some warmer sounds. "I have a whole collection of vintage keyboards, effects, guitars, and amps—a Vox AC30 and an old Fender Tweed Deluxe amp."

Schnier has been touring in support of *One* with New York musician Mitch Getz. "We're using Reason as the main accompaniment source," Schnier says. "I also have a Yamaha RS7000 workstation. It's so easy to rearrange the parts as needed and put in different loops that we could improvise over. You can do a lot of stuff on the fly. Nobody recommended [using Reason live], but we've been having a lot of luck with it so far. It's been great to hear [this music] through a house sound system. It's a whole lot different than headphones on the bus." ☻

For more information, contact Al Schnier; e-mail al@alschnier .com; Web www.alschnier.com.

One/Al Schnier

Jon Brion follows his muse.

Thinking Outside the Boxes

from the time he was a curious four-year-old banging on a garbage-can drum set to his recent Grammy nomination for the score of the motion picture Magnolia, Jon Brion has been busily exploring the infinite realms of music and sound—and usually pushing the envelope each step of the way. Never one to heed trends, Brion has opted for the narrower path, pursuing only his own muse. Judging by his successes to date—as a session player, a songwriter, a producer, and an engineer—his muse has clearly appreciated the attention.
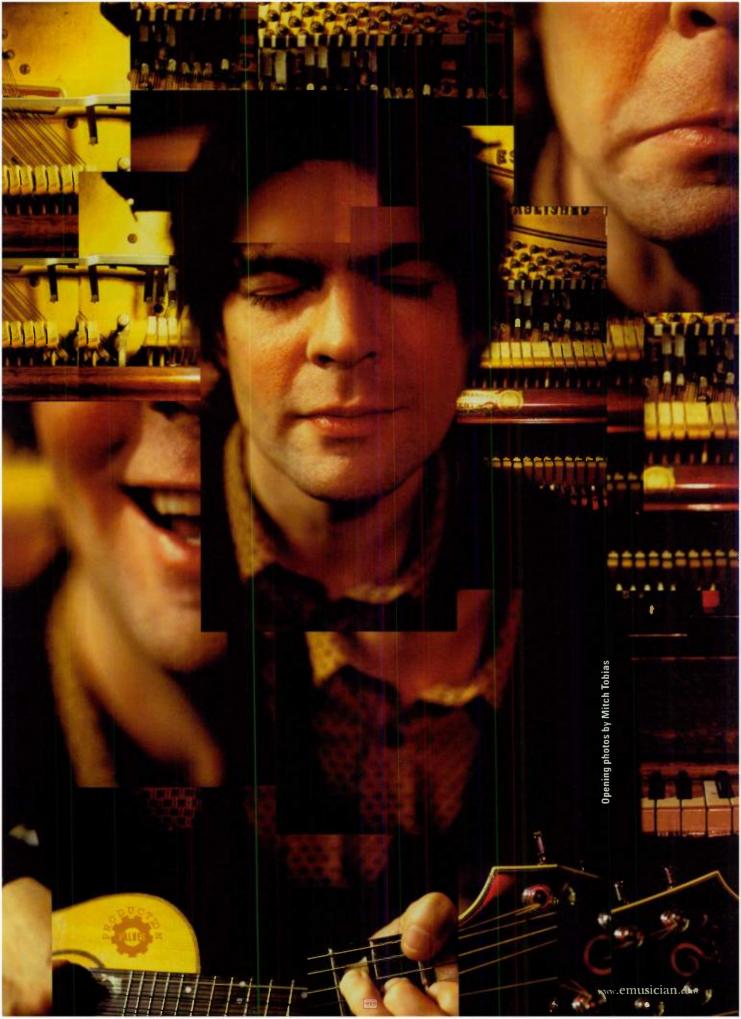
Brion's iconoclasm, innate musicality, and consuming quest for original sounds, not to mention his wonderfully diverse collection of unusual and vintage instruments, processors, and whatnot, have helped establish him as one of the world's most respected musician-producers. His production credits are as eclectic as any music geek's record collection, yet they read like a listing of recent Rolling Stone covers—Beck, Fiona Apple, Macy Gray, John Hiatt, Robyn Hitchcock, Dave Navarro, Elliott Smith, the Wallflowers, Sam Phillips, and the Crystal Method are but a few of Brion's satisfied clients. When studio veterans such as Bob Rock and T-Bone Burnett hit an overdub wall, they call on Brion, the one-man wrecking crew. His highly regarded session work (guitar, piano, vibraphone, harp, Chamberlain, and Optigan) on Fiona Apple's debut record, Tidal, stands as a prime example of his musical multitasking abilities.

Most recently, Brion's career as a producer has eclipsed his formidable work as a session player and songwriter. Indeed, it was the fruitful working relationship he developed with Apple that propelled him to the rank of producer on her second LP, When the Pawn . . . . Increasingly, Brion's career trajectory is reminiscent of eclectic musician-producers such as Brian Eno and Todd Rundgren: just as Brion has no shortage of odd instruments and inspiration, he has no shortage of opinions on what makes a recording compelling.

By Kenneth A. Woods

Thinking Outside the Boxes

What are you working on now?

I'm finishing the score for Paul [Thomas] Anderson's new movie, Punchdrunk Knuckle Love.

What are you doing to mix things up on the score?

We're doing something tomorrow on an old wire recorder.

Do you think lo-fi or 4-track recordings have value?

I think they're amazing. I've tried to learn from my experiences and other people's experiences of having a great piece of music on 4-track and then going to the studio and killing it or worrying too much. The 4-track cassette recorder performs a wonderful task in getting people to feel less self-conscious. Its purpose is home recording. Everything about it makes you not worry about the recording—which, in truth, is how you need to be when you're in the recording studio. I like to think of what I do as a 4-track approach, but with extra tracks.

The reason I can be such a hard-ass about my choices for mics and preamps is that I don't want to be equalizing when I'm working. I don't ever want to sit and watch a drummer hit a snare

drum for an hour. I get massively depressed if I do that. That's one of the great lessons of the 4-track—don't get so hung up on fidelity. People with a 4-track take a Shure SM57, shove it in front of a guitar, and say, "Go." The immediacy of that is so good, and that's what all recording should have—immediacy.

Is home recording relevant only when you're doing demos or going for a lo-fi thing?

Here we get into the realm of the D word, which is demo. I don't believe in demos. I think they're really bad. What happens is people spend their first-take energy on the 4-track—the one moment when they aren't self-conscious with the song. Then they go into the studio to try and recapture the magic. It's never going to happen. You're not the same person you were the day you recorded the demo on your 4-track. Now you're in a big studio and you're paying for it. It's not the 4-track sitting on your kitchen table. Everything's different now.

Every time you record a song, you should think of it as a version of the song. If you're dissatisfied when you're done, try doing another version—but remember to take a different approach. After all, you were already dissatisfied with the song.

What are the key benefits of digital recording in the personal studio?

The beauty of digital is that instead of wasting that wonderful moment when you're first messing around with something, you can capture that first-take magic with the same sort of immediacy as with the 4-track and with the same ease of operation, but with better fidelity. That unself-consciousness is the main thing I try to capture on recordings. Always keep things rolling.

FIG. 2: Among Brion's most used pieces of gear is his pair of MGM Church Cinemikes, one of which is shown here.

Try to get the moments when someone stumbles onto something.

Of course, digital is unbeatable for editing. There are times when I choose Pro Tools simply because of noise-floor issues, especially with ballads. I usually run Pro Tools in tandem with my analog setup.

What do you think are the most frustrating things about recording at home, and how can we avoid them?

The most frustrating thing is that computers are so powerful that they set up unreasonable expectations. Because they can do so much so fast, it makes you think you can just go and go. But crashes are virtually inevitable. You get a large number of tracks in the computer, and then it just goes kablooey. It's a number of tracks you could never expect to record onto an analog machine. Before digital I would never have kept all seven passes of the same guitar part. Now I'm asking my digital recorder to do things I never would have expected from an analog machine. Then you run around the room cursing the computer when it crashes. You can easily avoid all that stuff by remembering it's your job to be a good musician still. Don't get caught up doing endless permutations of parts.

How did you learn to operate your personal-studio equipment, and how long did it take?

I'm still learning Pro Tools. I'm not

FIG. 1: To come up with distinctive sounds for different songs, it helps to have various instrument collections on hand. Shown is an assortment of Brion's snare drums.

Thinking Outside the Boxes

very good with computers. When I'm working on real projects, I have somebody do all that stuff for me.

You're between personal studios right now. What are you looking for in a personal-studio location?

A gigantic room is what I want. I have a lot of bizarro drum sets and keyboards that I'd like to be able to have set up all the time. That way I can easily play an instrument without any setup, and I can pull microphones back and get the air around things. I'd like to have six upright pianos around. You can rent a truck on a weekend and pick up four or five amazing pianos—they're in the paper for free every week. For example, you could have one with an old-fashioned John Cage treatment, another with the hammers cut off and replaced with something else, another that's detuned honky-tonk style, or whatever. You could have endless great keyboard sounds on your records that nobody has heard before. I guess what I want is a laboratory.

What advice would you give to someone who is building a personal studio?

Buy one microphone that you really like. Just buy one really good mic and use it on everything: drums, acoustic guitar, vocals. Also, go to music stores and buy the stuff that nobody else wants. I go in and say, "Where's your junk?" Often you're better off buying a cowbell at a thrift store—and having a box of that kind of stuff—than you are buying an expensive piece of gear. I have a pile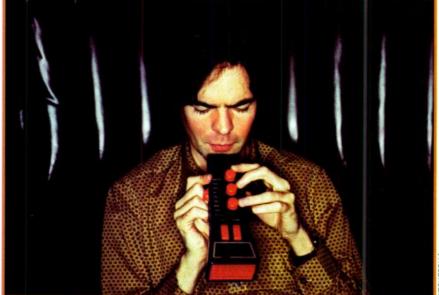 of cheap snare drums [see **Fig. 1**]. It's also great to have, like, ten tambourines when you're working on a track: one might hide behind things better; another might do a better job of poking through tracks that are really full; yet another might blend in just right and sound like it's part of the drum kit—you don't even sense it as a tambourine.

I also think comfort is important in a personal studio. The studio should be like you. If you like a big unholy mess, then let it be that. If you want good sounds, there's no faster way than recording in your bathroom. Every bathroom has a different sound. Even if you have a tiny practice amp in your bathroom, it's going to sound great. People need to get back into recording the molecules around a sound,

Brion is a big fan of sound-making toys and gizmos. Here he readies to play an instrument that's called the Gizmo.

MITCH TOBIAS

Thinking Outside the Boxes

the air. There's so much distinctiveness there.

Also, when you're recording at home, don't get hung up on what the standard of tone is. I think that notion should be abandoned. Even if you spend days, you won't get as good a tone as what a great studio can. So don't waste your time tweaking to get "record-quality" sounds. If you sit down with a song and spend an hour on a guitar sound, your brain will be gone.

Just get a quick, accurate version of the song. If you're going to tweak the tone, it should be for fun—to create original sounds. Why spend all day equalizing to get a B+ sound when you can just move on and get an A+ performance instead? There's a really clear hierarchy for me: first is the material, second is the performance, and last are things like sonics. It's great to have an A+ sound, but that's always secondary to the song and performance.

What are some of your secret weapons? Do you try to avoid doing the same things twice?

My secret weapons are secret, and I try not to use them all of the time so they won't be a crutch. [Laughs.] My secret weapons were developed by not doing the same thing all the time—and by remembering that that principle works best every time.

So I guess you always do the same thing.

Yeah—I always try something different. [Laughs.] Now it's not a secret.

What's the most amazing technique you have learned in the studio?

I think it's probably understanding the psychological environment that's important to making good music. As a session player, I've been able to examine a lot of producers' techniques, and I've experienced what it feels like to be told certain things. I've learned a lot about how to be in the right place mentally to record. The best tricks are more about attitude.

As far as mixing goes, I've watched Bob Clearmountain mix, and you think he's going to have all these secret boxes. But in fact, there aren't any. He sits with the vocal twice as loud as the rest of the

mix and listens to the song for hours. Each fader moves imperceptibly. By the fourth hour, everything sounds amazing—and he's been using only the faders. The real secret of the people who make great records is that they know when to stop and not worry about things.

What is the most important piece of gear you own, and why?

Me. Whatever the hell it is that makes me me and makes me want to find new things. I didn't realize that until I was telling someone that my best asset was my collection of vintage gear, and the guy said, "No, dummy, you are." It seems like a really obvious, dumb thing, but the real solution is realizing that everything you have is your palette. It's all important: your good mic, your bad mic—anything that makes a sound is potentially important. The various dishes and silverware from this room are just as vital as the most beautiful snare drum, because things are always changing—it's a new song and a new day. Things are constantly morphing. The real job at hand is to ride with it.

What pieces of your gear get the most use?

My MGM Church Cinemikes [see **Fig. 2**] and probably my pump organ or Chamberlain. My Dictaphone is big, too, because it gets used so much in my day-to-day life, which spills over into the studio. It's with me every step of the way, from the core idea of a new song to an interesting sound I hear to being an instrument that I play while singing into it and applying the speed control as an octave device. I've used it to make basic tracks that I then lay down on my multitrack machine. I do so many things with my Dictaphone.

What do you think of some of the new digital products such as modeling amps and vintage plug-ins?

The good thing is they're inexpensive, and they put a ton of tones in more people's hands than before. That's great. They open people up to playing with a variety of tones when recording. That's great, too. But when people buy a new product with a choice of 99

JON BRION: A SELECTED DISCOGRAPHY

Fiona Apple, *Tidal* (Sony/Work, 1996); guitar, piano, tack piano, harp, marimba, vibraphone, dulcitone, Chamberlain, Optigan

Fiona Apple, *When the Pawn*... (Sony/Work, 1999); producer, mixer, instrumentation

Eels, *Beautiful Freak* (DreamWorks, 1996); producer, engineer, guitar, trombone, Chamberlain

Eels, *Electro-Shock Blues* (DreamWorks, 1998); organ, Chamberlain

Macy Gray, *On How Life Is* (Epic, 1999); synthesizer, guitar, piano, marimba, orchestra bells, Chamberlain

John Hiatt, *Little Head* (Capitol, 1997); vibraphone, Chamberlain

Robyn Hitchcock, *Jewels for Sophia* (Warner Brothers, 1999); organ, bass, percussion, drums, vocals, producer, engineer, Chamberlain, keyboard bass

Aimee Mann, *Whatever* (Geffen, 1993); producer, arranger, organ, bass, drums, glockenspiel, electric guitar, keyboards, tambourine, vocals, Mellotron, Chamberlain

Aimee Mann, *I'm with Stupid* (Geffen, 1995); acoustic guitar, electric guitar, bass, harmonica, percussion, cello, drums, keyboards, vocals

Aimee Mann, *Bachelor No. 2* (Supergo, 2000); producer

Miranda Lee Richards, *Herethereafter* (Virgin, 2001); bass, guitar, bells, producer, engineer, Chamberlain, Marxophone, Optigan

Elliott Smith, *XO* (DreamWorks, 1998); Chamberlain

Rufus Wainwright, *Rufus Wainwright* (DreamWorks, 1998); producer

The Wallflowers, *Bringing down the Horse* (Interscope, 1996); guitar

Thinking Outside the Boxes

sounds, they spend the first night going through all 99 sounds. Then they pick a couple they like, and that's basically what they use the rest of the time, because the other sounds aren't actually that different.

In every modeling box, there's a device that creates distortion, and it has its own "grain" or window of fidelity. The unit may claim to have every sound imaginable, but it's still all those sounds within that grain. The fact is you always get a more individualized sound from boxes devoted to a single function. So if you've already used a certain fuzz box on a song, plug in a different fuzz. Now your song has two entirely different grains of distortion, which gives that much more tonal "life force" to the track.

Another trick is to combine your modeling boxes with other effects. One of my favorite sounds comes from patching a factory preset on my Line 6 Echo Pro delay modeler through this little Ibanez FZ7 fuzz box. [For more on the Echo Pro, see the review on p. 146.] There's a preset on the Echo Pro that's delay with a chorus [Digital Delay w/Mod], and when you put it through this cheap little Ibanez fuzz box, you get this great sound from the fuzz being driven like crazy. There's all this different information from the delay time moving and the chorus. You get the sickest, meanest fuzz sound out of it.

Another tip is to run your Line 6 Pod through an old cassette deck to keep it from sounding like a glorified Rockman. You'll turn it into a more important piece of a much more complex sound.

You can take the same approach with samples. Instead of using samples of vintage instruments, why not take a woodwind sample or something with some complex harmonics, run it through a fuzz box and then into a guitar amplifier, mic it from a few feet back, and com-

press it so you can hear a little bit of the room? Then you have something with complex harmonics making the fuzz box go crazy. It can't handle how complex the initial tone is, so it starts spitting out weird stuff, and that's going to an amp and getting colored by the tubes, then that's getting pumped into a room and it's moving molecules, and you're recording the reflections of actual atoms moving. With compression you can decide how much of the information slapping around the room you want.

What other tips do you have for overcoming the limitations of plug-ins and presets?

Don't always use the plug-ins. Run out of your I/O and into some hardware. Most people think their Neve, LA2A, or whatever plug-ins sound great. But most of these people have never even seen a real Neve, let alone listened to one. Stop expecting plug-ins to sound as good as the devices they are simulating and respect them for the good sounds that they do make. Mix it up. Go to the local used-instrument music store, spend a hundred dollars on some weird effects pedals, and experiment with those as your outboard effects. I'm more interested in acoustic ambience devices and making use of acoustic environments the way engineers did in the old days. Joe Meek used to have a speaker with a fan in front of it and a mic on the other side.

Here's something else you can try: I heard that when Trent Reznor starts a project, he makes a list of things you're supposed to do, such as eighth-note tambourine on the chorus and chorus bigger than verse. The list gets posted in the studio, and you're not allowed to do any of those things. That's a great way of working.

Do all the different recording mediums and techniques somehow complement one another and influence the material or the artist?

Definitely. I use different mediums a

An overview of Brion's amplifier collection suggests a fondness for tubes.

lot to consciously color what I'm working on. A specific medium can force everyone in the environment to work differently, especially the players in the room. Pro Tools, for example, makes people play a certain way. You stare at the screen and you know that you can play ten tracks of overdubs, and someone will comp it together when you leave.

People playing live together results in something entirely different. Tell everybody, "Okay, we're going live," and see what happens. I find that people perform really well live to two tracks. It gives you some of what we had with bands like the Beatles and the Rolling Stones.

We need more communal music making, human beings responding to each other. Now there's the overdub, and it has become massively powerful with digital. Recording as an art form is about using all these different mediums as a palette, not an instruction manual. Potentially, everything you own is the perfect thing for the job. No one piece of gear, technique, or scenario is any more important or useful than another. In fact, your most prized piece of gear or your best friend's secret weapon can often be the wrong thing for a given song.

Kenneth A. Woods is a writer, producer, and musician living in Los Angeles. He is working on a book titled Rock and Roll Is Dead.

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Loop-a-

BY TODD SOUVIGNIER

Welcome to **EM**'s Loop-a-palooza, a dual-platform time-shifting spectacular. We've rounded up the best new software tools for loop-based music production and put them to the test.

Descended from the MIDI sequencer and the digital audio sequencer, multitrack loop sequencers represent a third generation in the evolution of sequencing software. By making it possible to easily tempo-match and layer audio loops, loop sequencers have already changed the way thousands of artists make music. Although the programs covered in this roundup are often surprisingly powerful and versatile, they all center on the lowly loop.

LOOPOLOGY

For the purposes of this article, I'll define a loop as an audio file or a section of an audio file that is designed to play repeatedly. Loops are often short recordings of rhythm parts and are used as "building blocks" within larger arrangements. When a loop is edited correctly, it produces a smooth, continuous pattern. If the editing is sloppy, you'll hear a pause, an increase in speed, a hiccup, or some other anomaly.

When people first started working with samplers and digital audio sequencers, they naturally tried to layer different audio loops. That often proved to be problematic. If two loops aren't the exact same tempo and the exact same length (or an even multiple or fraction thereof), they just won't lock. The loops must match, and match well, or they will quickly drift apart.

illustration by DAVE EMBER

palooza!

A giant-size

survey of the best new

software for loop-based

music production.

Loop-a-palooza

SLICE AND DICE

The programs in this roundup solve the loop-matching problem by viewing each audio file as a series of individual notes, drum hits, or segments typically referred to as slices. First popularized by Propellerhead ReCycle and Sonic Foundry Acid, loop slicing lets you change the tempo and duration of a loop while retaining the original sound and pitch of the music.

Slicing doesn't really alter the sounds of the instruments. It just chops each loop into a series of notes and plays the notes at the designated time. When you slice up an audio file, you create an edit list or playlist that is tied to the composition's tempo.

That's a better approach than using digital-signal-processing (DSP) time-stretching algorithms to expand or contract loops. Aside from their other limitations, time-stretching algorithms almost always color the sound and frequently yield unnatural results. In addition, they demand considerable processing power.

When you import an audio file into a loop sequencer, the program uses a process called transient detection to determine where the beats are. The pro-
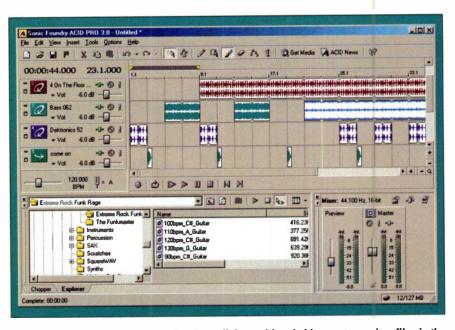
FIG. 2: Sonic Foundry Acid was the first loop-slicing multitrack. It's easy to preview files in the browser view and then drag the files up into the track display for arranging and processing.

gram slices the file into segments, placing a marker before the major transients (peaks).

Some programs also superimpose a tempo grid, placing a slice marker at note intervals (such as every quarter note, eighth note, or 16th note). Markers can usually be adjusted manually, but you rarely need to touch them—the automatic marker placement is generally dead-on in all of the programs surveyed here.

Keep in mind that in most cases, the audio file is not actually being segmented. Rather, the program creates and modifies a list of the markers, pointing to time positions within the original audio file. It's that list of marker times that gets manipulated when you change the tempo. Slicing not only makes it simple to change the tempo of loops but it can also let you quantize loops, derive groove templates from audio files, and perform wholesale drum and instrument replacements. (These programs also offer separate pitch-shifting functions so you can prop-

erly match pitches as well as tempos.)

Think of the loop as a kind of musical building block consisting of the original WAV or AIFF audio file along with the slice marker list and usually other information, such as the detected tempo or assigned key. Manufacturers use proprietary technology to implement this shiftable loop package; the REX2 and Acidized formats are supported by some of the originators' competitors. As you'll see, the need for preformatted samples is starting to disappear. (See the table "Looper Specs" for a list of supported file formats and other details.)

Most of the programs in this roundup are available for PCs with Pentium processors running most of the recent versions of Windows, including 98, ME, 2000, and XP. Three of the programs run on Power Macs with Mac OS 8.6 or higher.

This article omits a number of me-too or junior-edition programs, and it centers on the middle and upper ranges of the price and functionality spectrum. (See the table "In Review" for a list of the programs that have received full reviews in EM.) All the programs in this roundup are multitrack layering environments with the exception of ReCycle, the one that started it all. (You'll
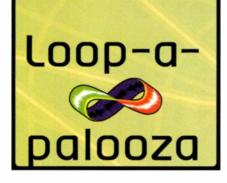
FIG. 1: Propellerhead ReCycle pioneered the loop-slicing concept; the vertical slice-marker lines in the waveform display can be created automatically or by hand.

Loop-a-palooza

also find an alternative approach to loop manipulation in the sidebar "Ain't That a Groove.")

RECYCLE (MAC/WIN)

Propellerhead Software (maker of Re-Birth) started the loop-slicing madness back in 1994 with the original ReCycle. The **EM** editors liked version 1.0 so much they gave it an Editors' Choice award the following year. ReCycle 2.0, released in 2001, keeps this venerable tool relevant in the 21st century.

Designed as an adjunct utility for audio-editing and multitrack software, ReCycle pretty much does one thing: it offers better time shifting through loop slicing—and it slices loops exceedingly well (see **Fig. 1**). Propellerhead has addressed a number of the original version's shortcomings—for example, ReCycle now handles stereo files—and the company has kept the

program focused, fast, and simple.

ReCycle lets you adjust the tempo of any loop and export it as a new WAV or AIFF file. Make sure you remember to select Transmit as One Sample from the Process menu when you want a single file. Otherwise ReCycle's default settings will cause it to save each segment in the loop as a separate audio file. If you have a sampler, you can then load the slices into a new preset and trigger the slices using any sequencing software. ReCycle even exports a Standard MIDI File beat map for that purpose.

Once a groove has been broken down to the component drum hits and their sequence of Note On times, you can easily adjust the tempo, quantize the performance, or replace any of the sounds.

ReCycle's proprietary REX2 files can be imported as a single audio track into a REX2-compatible sequencer (such as Reason, Emagic Logic Audio, or Steinberg Nuendo or Cubase VST) and

FIG. 4: In the Sonar and Plasma Loop Construction view, Beat Grid size and Transient Detection sensitivity are controlled by the two sliders above the waveform display.

played back with control over the tempo. According to Propellerhead, REX2 support in Mark of the Unicorn (MOTU) Digital Performer is forthcoming.

ReCycle supports direct SCSI transfer for major hardware samplers, and it offers MIDI sample-transfer capability and special Akai-format support. (ReCycle is the only program in this group that provides specific sampler support.) The software includes a handful of basic DSP processes, such as normalization, DC-offset correction, real-time EQ, compression, and pitch shift.

ReCycle ran without a hitch during my tests. If you've ever used sound-editing software, it'll take you ten minutes or less to figure out how to work with the program.

ACID PRO (WIN)

Sonic Foundry Acid's innovation was to deploy ReCycle's loop-slicing paradigm on a multitrack level. The program has been through a couple of overhauls since its 1998 release, and Acid Pro 3.0 (see **Fig. 2**) retains its position at the head of the pack.

Acid Pro lets you drag time-shifted loops, MIDI files, long digital-audio files, and even a video file into an arranging window with one file per track. In that respect, it's different from other audio multitrack programs, which typically let you put multiple files into a single track. With Acid Pro, you end up having more tracks, which could be a detriment when adding real-time effects.

New in Acid Pro 3.0 is the Beatmapper, which handles song-length files, dropping markers and controlling tempos across long recordings. MIDI playback and recording is another big new

FIG. 3: You can drag files from Cakewalk Sonar's browser window (foreground) into the track-arranging display (background); Plasma looks almost exactly the same.

Looper Specs

PRODUCT NAME	RECYCLE 2.0	ACID PRO 3.0	PLASMA 1.0	SONAR 1.3.1	FRUITYLOOPS 3.4	PHRAZER 1.0.2	LIVE 1.1
Platform	Mac/Win	Win	Win	Win	Win	Mac	Mac/Win
Multitrack Audio Editing	no	yes	yes	yes	no	no	yes
Audio-File Editing	yes	yes[1]	no	no	no	no	no
Edit Loop Start and End Points	yes	no	no	no	no	yes	yes
Time Stretching (DSP)	no	no	no	no	yes	no	no
Preview Time Shift on Processed Loops	n/a	yes	yes	yes	no	no	yes
Preview Time Shift on Any Loop (from file browser)	no	yes	no	no	no	no	yes
Multiple Loops per Track	n/a	no	yes	yes	no	yes	yes
Audio Recording	no	yes	yes	yes	no	yes	yes
MIDI Recording	no	yes	yes	yes	yes	no	no
MIDI Editing/Playback	yes[2]	yes	yes	yes	yes	no	no
MIDI Controller Support	no	no	yes	yes	yes	yes[3]	yes
MIDI Resolution (ppqn)	n/a	24,576	960	960	768	n/a	n/a
Multiple Undo	no	yes	yes	yes	no	no	yes
Step Sequencer	no	no	yes[4]	no	yes	no	no
Piano Roll	no	yes	yes	yes	yes	no	no
Event List	no	no	yes	yes	no	no	no
Alternate Meters (such as 5/4)	yes	no	yes	yes	yes	yes	yes
Audio Volume Automation	no	yes	yes	yes	yes	no	yes
Effects Parameter Automation	no	no	yes	yes	yes	no	yes
Built-In Synths	no	no	yes	yes	yes	no	no
DirectX Plug-In Support	no	yes	yes	yes	yes	no	no
DXi Support	no	no	yes	yes	yes	no	no
VST Plug-In Support	no	no	no	no	yes	no	yes
VST Instrument Support	no	no	no	no	yes	no	no
ASIO Support	yes	no	no	no	yes	yes	yes
ReWire Support	no	no	no	no	no	yes	yes
DLS Support	no	yes	no	no	yes	no	no
File Format Support	WAV, AIFF, SDII, MIDI	WAV, AIFF, WMA, MP3, RA, MOV, AVI, OGG	WAV, AIFF, WMA, MP3, RA (export), MOV, AVI, MIDI	WAV, AIFF, WMA, MP3, RA, MOV, AVI, MIDI	WAV, MP3 (save only)	WAV, AIFF, SDII, MOV	WAV, AIFF
24-bit, 96 kHz Support	no	yes	no	yes	no	no[5]	yes
Included DSP Effects	no	18	6	23	24	12	10
Live Input Effects	no	no	yes	yes	no	no	no
Aux Channels	no	26	no	16	16 FX channels, 4 sends	2	4
Video Track	no	yes	yes	yes	no	yes	no
Price	$179	$499	$49	$739	$139 with manual; download version $99; BeatSlicer $35	$299	$299.95; $349 from Ableton Web site

Notes:
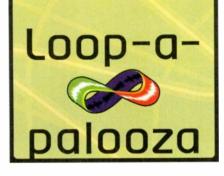
1. Includes Chopper tool that allows nondestructive selection from larger files.

2. Exports MIDI sequence that can trigger sampler and slices.

3. Gate tracks on and off from MIDI or from QWERTY keyboard.

4. Includes Fruityloops Express light edition.

5. Supports 16-bit, 96 kHz audio.

Loop-a-palooza

feature. It's quick and simple to configure the program, start recording, and edit in the Piano-Roll view.

Also new in 3.0 is DLS (Downloadable Sound) playback support and video scoring. AVI and MOV (QuickTime) files open as a "filmstrip" in the Track view. The strip display resizes nicely; zoom in far enough, and you can spot audio cues with frame accuracy. There's also an optional video display in a small dedicated window, and Acid Pro supports external video displays, which is really the way to go when syncing sound to picture.

Unfortunately, you'll be a bit disappointed if you try to import an MPEG video into Acid Pro. A dialog box appears and states that a plug-in purchase is required. And forget about trying to edit the video strip in any way from within the program. When you click on the video track, built-in ads for Vegas Video and Video Factory pop up.

Any skimping in the video department, however, is in marked contrast to the audio side of the package. The

Sound Forge XP Studio 5.0 audio-editing software and the Vegas Audio LE multitrack program are bundled with Acid Pro. The package also includes CD-ripping and -burning capabilities and 18 DirectX audio plug-ins.

When it comes to looping audio, Acid Pro is king. You can just open and play any session, go into the Explorer display, click on Auto Preview, and then single-click on any audio loop in your collection. Better yet, use the arrow keys to scroll quickly through whole folders of loops.

You'll notice that loops are previewed in sync with the session. If they're cut correctly, loops that haven't been previously Acidized are previewed at the session tempo. That's an important point: Acid Pro 3.0 slices and tempo-shifts loops before you even import them into the program! That's what I call performance. Acid Pro anticipates your needs and graciously provides the solution.

Acid Pro has multipurpose faders in the Track display as well as a Mixer window that displays all channels, buses, and effects. Volume, pan, and effects sends can be automated and

FIG. 6: Creating new sliced-audio loops for Fruityloops requires Zero-X BeatSlicer, a separate purchase.

edited using conventional break-point envelopes.

I discovered an undocumented feature in Acid Pro: Paint Brush erase. With the Paint tool selected, you can right-click on the Track display to delete events. It's faster than switching to the Eraser and works only with the Paint tool, not the Pencil tool. It's a little quirky, but it's convenient.

Acid Pro 3.0 has only a few shortcomings. As mentioned previously, it's strictly a one-file-per-track program. There is no support for alternate meters such as 5/4 or 7/8, and it notably lacks effects-parameter automation, although it does let you automate effects-send levels. Acid Pro doesn't have DXi or VST Instrument support. Also, Acid Pro's graphic-editing tools are "modal"; you'll find yourself constantly having to switch between the Pencil, Paint, Selection, and Eraser tools. As trade-offs go, however, those are a small price to pay for the outstanding looping performance, easy audio and MIDI recording, overall ruggedness, and sensible layout of Acid Pro.

PLASMA AND SONAR (WIN)

At a suggested retail price of $49.95, Plasma is Cakewalk's attempt to grab the low end of the looping market, and it competes vigorously with Sonic Foundry's Acid Music junior edition. Plasma delivers the important bread-and-butter features, and it also provides sufficient power to get some real work done.

Plasma looks and works a lot like Cakewalk's high-end Sonar sequencer (see Fig. 3). From the user-configurable

FIG. 5: The Image-Line Fruityloops step sequencer (center) can drive sliced loops, samples, and MIDI sequences. The Fruity Slicer plug-in (on the right) is for loop playback.

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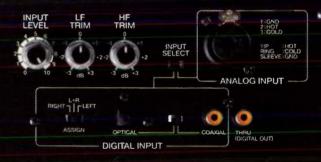
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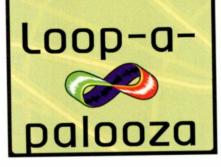
toolbar right down to the identical windows and dialog boxes, Plasma is "Sonar Junior" with an emphasis on looping features.

Plasma and Sonar both feature a Loop Construction window that is highly reminiscent of ReCycle (see **Fig. 4**). Instead of offering one slider, Plasma has two: you can generate markers based on transient detection or simply apply a grid of note-length intervals. Creating new markers by hand requires a fast double-click, which is trickier than it sounds.

Whereas the default behavior of Acid is to slice files without asking, you have to tell Plasma and Sonar to slice each time by clicking on the Enable Looping button in the upper-left corner of the Loop Construction window. Cakewalk representatives say that's necessary because Sonar doesn't deal exclusively with loops. You can also import other audio data into projects. Be that as it may, it's just not as slick from a loop-centric perspective.

Once a file is sliced and looped, it's easy to "paint" blocks across the track window and start building an arrangement; there are no confusing drawing tools in Plasma.

The Loop Explorer view appears in Plasma and Sonar as a simple floating window. It lets you sort, view, and audition any supported audio file. The Loop Explorer's Auto Preview function plays audio files that are not yet part of your session as you click on them in the display. You can even preview as many as 16 loops simultaneously!

If the selected audio file is a preprocessed "Groove" or Acidized clip, the file will time-shift and preview in the current project tempo and key. If the file is simply a new WAV or AIFF file, it previews at its original tempo, out of sync with the session. Unfortunately, that's not quite at the level of the previewing found in Acid Pro (or Ableton Live, which I'll cover later).

Cakewalk's extensive experience in developing MIDI sequencers really shows in Plasma and Sonar. MIDI and audio production require a deep and complex tool set, and neither Plasma nor Sonar skimps on the features. Nevertheless, it's as easy to sequence in Plasma or Sonar as in any program that I've ever used. Both programs offer unlimited audio and MIDI tracks, video import/export, support for automatable DirectX 8 effects plug-ins, support for alternate meters (Brubeck, here we come), unlimited layers of Undo, and fairly straightforward multitrack audio editing and automation. Moreover, both packages come with lots of DSP effects plug-ins and loop libraries.

Any gripes I have about Sonar or Plasma are essentially philosophical. For example, Plasma and Sonar rely on the current-time cursor position as the starting point for paste operations instead of using the edge of the current clip or track selection. That forces you to reposition the cursor before

FIG. 8: The Munge menu is revealed by clicking on the wrench button beneath Phrazer's Track view.

every paste. It can slow things down and seems counterintuitive.

Nonetheless, Plasma and Sonar provide everything that you would typically expect in a loop sequencer and a whole lot more. They're feature-rich and largely well-designed products capable of producing high-quality professional works. Though certain parts of Plasma and Sonar (such as basic MIDI recording) are dead simple, other aspects may have you scratching your head at first. By the time you figure out basics such as the toolbars and Channel Strip, however, you'll be cruising along.

Some aspects of the looping implementation in Plasma and Sonar may not be as slick as Acid Pro and Ableton Live. But by including things such as automatable plug-in effects, multiple files per track, alternate meters, and DXi synth support, Plasma and Sonar offer much more complete sequencing tool sets.

FRUITYLOOPS 3 (WIN)
Developed by the Belgian company Image-Line Software, Fruityloops 3 is a self-contained production environment for electronic dance music. Centered on a step-sequencing window (see **Fig. 5**), Fruityloops is not really an audio multitracker or a MIDI sequencer; it's more of a post-ReBirth groove machine on steroids. "Techno playhouse" might be an apt description.

Image-Line packs a lot into the step-sequencing metaphor, and not just

FIG. 7: BitHeadz Phrazer was the first multitrack loop sequencer for the Mac. The sliders at the bottom of the screen are for a compressor effect.

Loop-a-palooza

drums and bass-line generators. You'll also find soft synths galore, sample playback, and time shifting. Composing in Fruityloops entails assigning sounds to the step sequencer and setting up patterns that are then arranged in the Playlist window. One of the possible sound sources is a sliced-up loop.

I worked with the full version of Fruityloops, but a scaled-down Pro version is available, too. That package lacks automation, ASIO support, and a piano-roll display. Fruityloops represents the modular approach to software design; things such as editors and additional soft synths are available as separate purchases. The same is true of time stretching, which is not built in to Fruityloops

as an integral central feature the way it is in Acid and Live.

An additional $35 purchase, Zero-X BeatSlicer is needed to import, slice, and save loops (see **Fig. 6**). BeatSlicer provides a rudimentary ReCycle-style slicing display, offers an assortment of tools for handling loops, and interacts with Fruityloops.

Grooves exported from BeatSlicer are laid out in steps as in ReCycle, one segment per step-sequencer channel. The terraced output makes it incredibly easy to go nuts with drum-replacement surgery, allowing you to switch out every hit and still retain much of the original feel.

BeatSlicer works with another included Fruityloops plug-in called Fruity Slicer. It lets you split up any WAV file and lines up the slices in the step sequencer. A Fruity Slicer playlist takes up a single channel on the step sequencer and is much easier to manage than the sprawling BeatSlicer exports.

Fruityloops doesn't do audio record-

ing and multitracking in the conventional sense, and that's no mark against it; it's a different type of program. It has a straightforward and rather cool piano-roll editing interface and allows you to play MIDI sequences right into the beatbox. The MIDI setup was difficult to use, and I had to enlist the help of an Image-Line representative to get my keyboard and sound modules talking to the program.

The program has some really intriguing aspects, such as the Fruity Scratcher effect plug-in. (Sadly, that module is completely undocumented.) Certain controls, such as the channel mutes, are hard to identify. (Check for the tiny green lights along the left side of the step window.) In addition, most functions in Fruityloops are buried three layers down within submenus. It's not exactly a dive-in-and-use-now program.

Fruityloops makes several disorienting departures from typical Windows interface conventions, including unusual uses for the right mouse button. I also found some glaring errors in the manual; certain things have been relabeled since the last printing. That kind of unprofessional lapse might lead you to conclude that this is more of a grand toy than a workhorse production system.

Nonetheless, there is plenty to like about Fruityloops 3. For example, nearly all effects parameters are easily automatable by mouse or external MIDI controller. The program is full of nifty novelties such as a two-knob plucked-string synthesizer and the Beepmap graphic-to-audio plug-in.

AIN'T THAT A GROOVE

If you're looking for an approach to loop manipulation other than those found here, check out the sample libraries from Ilio Entertainments (www.ilio.com) and Spectrasonics that employ Groove Control. Groove Control is similar to what you get when working with ReCycle, but the loop slicing has already been done for you. That means you just load a Groove Control–enabled loop into your sampler, open the Groove MIDI tracks in your sequencer, and get down to business.

I'd been working with the audio version of Ilio's *Ethno Techno* sample library, painstakingly ripping all the files to my PC, cutting the tracks into individual hits, and jumping through a number of hoops to tempo-match various samples. Although I've occasionally used Acid to do some of the tempo-matching for me, the number of steps in-

volved often outweigh the benefits. With the Groove Control version of the CD-ROM, I dumped the sample files into my Kurzweil K2500, assigned each patch to the proper MIDI channel, and loaded the associated Standard MIDI Files into my sequencer. Within a few minutes, all of the features of my sequencer—tempo shifting, quantizing, Velocity scaling, and more—became available to manipulate the sounds. And I never once cracked the manual!

Keep in mind that there is one major limitation with Groove Control: it works only with loops from Ilio and Spectrasonics. There are nine Groove Control–activated titles (starting at $199), covering a broad range of categories. Each CD-ROM packs hundreds of megabytes of samples, and new loop CDs are coming on the market all the time.

—*Dennis Miller*

Loop-a-palooza

The designers have thoughtfully put resonant filters nearly everywhere you turn, and the program places little strain on the CPU. It ran smoothly at all times.

I would prefer it if Fruityloops' loop-slicing features were more tightly integrated and the program didn't require the purchase of a second product from another company, although the interoperability between Fruityloops and BeatSlicer is quite reliable. According to the company, many people prefer Image-Line's modular approach to software.

PHRAZER (MAC)
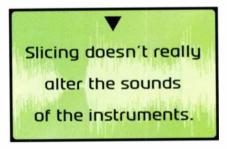
BitHeadz is best known for its beloved Retro AS-1 software synthesizer and Unity DS-1 software sampler. Released in 2000, Phrazer was for a time the only looping multitracker on the Macintosh (see **Fig. 7**). At this point, Phrazer could use an upgrade, although it delivers a basic Acid-type experience on the Mac platform and still does some pretty cool things.

At first glance, Phrazer is similar in layout to Acid, Sonar, and Live. It has a file-browsing window and a track display with an edit window. You drag files in, slice them, and arrange them, all under master tempo and key control. Phrazer lets you place any number of loops in a single track, which is preferable to the Acid one-loop-per-track approach.

Phrazer has no MIDI sequencing and fairly rudimentary audio editing but makes up for those shortcomings with interoperability: it includes MOTU Audio System (MAS) output and Free-MIDI input for MOTU Digital Performer compatibility. ReWire support allows Phrazer to route audio to a Steinberg Cubase or an Emagic Logic Audio track, and DirectConnect provides a comparable service for Pro Tools users.

Another nice touch in Phrazer is that MIDI notes and QWERTY keystrokes can gate tracks or trigger one-shot audio files. This good idea is also implemented in Ableton Live. One feature that is not seen in Live is MOV support. Phrazer lets you import any QuickTime movie, spot audio to the frames, and export the video with the new soundtrack. Phrazer also reads Acid 2.0 files.

Phrazer employs what Bit-Headz calls Munge dialog boxes and the Munge menu, which could be a source of confusion because the term is never really defined. Munge is a rather obscure and slightly derogatory term that usually means to imperfectly transform information. Munge can also mean to change data in some way that can't or won't be explained.

> ▼ Slicing doesn't really alter the sounds of the instruments.

In this case, it's Phrazer-speak for a "process" or destructive edit.

In Phrazer, most of the audio processes, including Crop and Normalize, as well as the loop tools are concealed beneath the Wrench button (see **Fig. 8**). It's too bad those basic features were not made more accessible through keyboard shortcuts or a redundant Process menu.

Other important tools are likewise hidden away. To insert a track effect or add an automation event, you must hold down the Control key while clicking on a Track display block—not the most obvious move. You can also get to the effects selection from the Edit menu by choosing Add Event. Inserting real-time effects is an important feature in any audio program; it merits
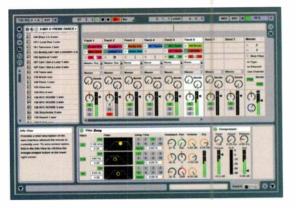
FIG. 9: Ableton Live's mixer is where most of the arranging work takes place. Filter Delay and Compressor effects are seen in the lower section; all parts are automatable.

some salient onscreen control or at least a menu option with the word *effect* in it.

Phrazer's built-in effects sound acceptable, but their controls are rudimentary, and I was quite surprised to find no effects presets or factory patches. You also won't find effects-parameter automation or VST plug-in support, things that most users have come to expect.

Speaking of omissions, there is no mixer automation editing using breakpoint envelopes. Phrazer does let you insert volume and pan "events" into the Track display, a somewhat unusual and rather counterintuitive way of mixing. In addition, you can't scroll the Track display by dragging a selection offscreen, and there is no way to relocate playback by clicking in the Track display; you must use the Transport palette.

When importing a new sample, Phrazer interrupts the process with three dialog boxes. The first asks if you want to enter split points. The next dialog asks if you want to "correct" the tempo of your loop. (The tempo must match the length and number of beats of the sample, and by its default settings Phrazer usually thinks it detects a mismatch.) Finally, the Sample Munge Tempo dialog appears, in which the user helps the program calculate the loop tempo.

One could argue that Phrazer is essentially a dedicated looping application. As such, it might have served users
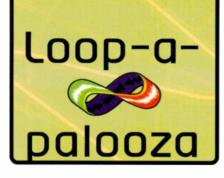
better by assuming that most imported audio files are in fact loops. It could then go ahead and slice them and calculate the darn tempo without human intervention. Ask About Tempo and Ask Default Split Points can be turned off. Unfortunately, deselecting those options doesn't get you automatic slicing and tempo calculation; it just suppresses the dialogs and makes you initiate slicing or calculation in every case.

Phrazer has two modes of slicing: transient detection and beat grid. The program provides slider control over slicing, and though the results are acceptably accurate, the implementation is nowhere near as keen as in ReCycle or Plasma.

Phrazer's PDF-only owner's manual

is a fairly good attempt. It includes several errors, however, and it's peppered with the kind of annoying typos that a spelling and grammar check could catch. I'd like to see a hard-copy manual with a few tutorials, some online help, an upgraded program, and a more streamlined looping procedure. If Phrazer were a low-cost program—less than $100—I would be less critical of it. But at $299, most Mac users will likely turn to Ableton Live instead.

LIVE (MAC/WIN)

Straight out of Berlin, Ableton Live is the only multitrack loop sequencer that runs on Mac and Windows. It's also a great-looking program, establishing a new style for audio software.

Smartly sidestepping the sequencer-feature wars, Ableton positions Live as a loop-sequencing instrument meant for real-time jamming. With the advent of underground laptop-computer music and a growing number of mainstream acts of every stripe taking computers onstage, live performance could well be a growth niche for the music-software biz.

Like Phrazer, Live lets you place multiple loops in each track and tracks in Live respond to MIDI Note On messages or QWERTY keystrokes. The similarities end there, however, because Ableton takes loop sequencing in a whole new direction.

Although Live has a track-arranging display, the program is centered on a mixer (or Session) window (see **Fig. 9**). What's unique is that you drag audio files into the mixer, stacking up as many loops, one-shots, or long files as you like in each mixer strip. You can play each clip by clicking onscreen Play buttons, or you can assign them to any MIDI note or computer key. Naturally, all

FIG. 10: Click on the Zoom tool—the blue bar just below the waveform display—to scroll and zoom within Live's Clip view.

looped audio is time shifted, so it stays in tempo with the session.

By de-emphasizing the Track view and block-arranging while featuring the straightforward, improvisatory Session interface, Live provides one of the fastest composition environments available. It combines file-browsing and effects-selection functions in one multipurpose panel along the left side of the screen.

The program has ten built-in effects, five of them variants on the delay line. Each effect is cleverly implemented, includes intuitive graphic controls, and has parameters that are fully automatable by MIDI controller or onscreen dragging. VST effects plug-ins are supported on Mac and Windows, but sadly, there is no DirectX plug-in support on the PC. There's also no MIDI sequencing and no real audio editing, although Live does allow you to record on the fly and to define a loop from within a larger file.

Although files can be dragged from the browser into the linear arranging display, the hip way to arrange is in real time. Turn on the record function and start firing clips you've loaded into the mixer; selections are recorded, and they can subsequently be viewed and edited in the Track window. All volume, pan, or effect-parameter changes are also recorded and played back; the conventional break-point automation envelopes can be edited in the track display.

Live includes several nice touches, such as zoom navigation. Rather than using a conventional magnifying glass, you click on a tiny file overview and drag to select a zoom range and center point (see **Fig. 10**). It's a unique and fluid way of zooming. (For a differing opinion about this feature, see Erik Hawkins's review of

Ableton Live on p. 116.) Another elegant refinement is Live's three file-browser views. Each can be set to a different root directory, sparing you from having to view the entire file structure.

Live includes detailed contextual help: hold the mouse pointer over any onscreen widget, and a complete description appears in the lower-left quadrant. To get rid of the Help display, click on the arrow in the lower-left corner.

High performance is the ultimate refinement. Like Acid Pro, Live lets you preview loops at session tempo, even loops that have not been presliced. You can let your session roll and then arrow-key through Live's file browser to audition batches of loops in time with your composition. As long as the loops have been cut correctly, they'll sync.

Windows users will notice that no functions are assigned to the right mouse button; Live's PC version mirrors its Mac twin in nearly all respects. I wish

Live supported more file formats—such as SDII, MP3, or Red Book (CD) audio—but with conversion tools readily available elsewhere, the lack of built-in file support is by no means an insurmountable problem.

Other programs allow more control over the slicing process, letting users independently move or manually create slice markers. Live is a little more restrictive in that regard and a little more automated in its slicing; in practice - you are not likely to need or miss the extra work.
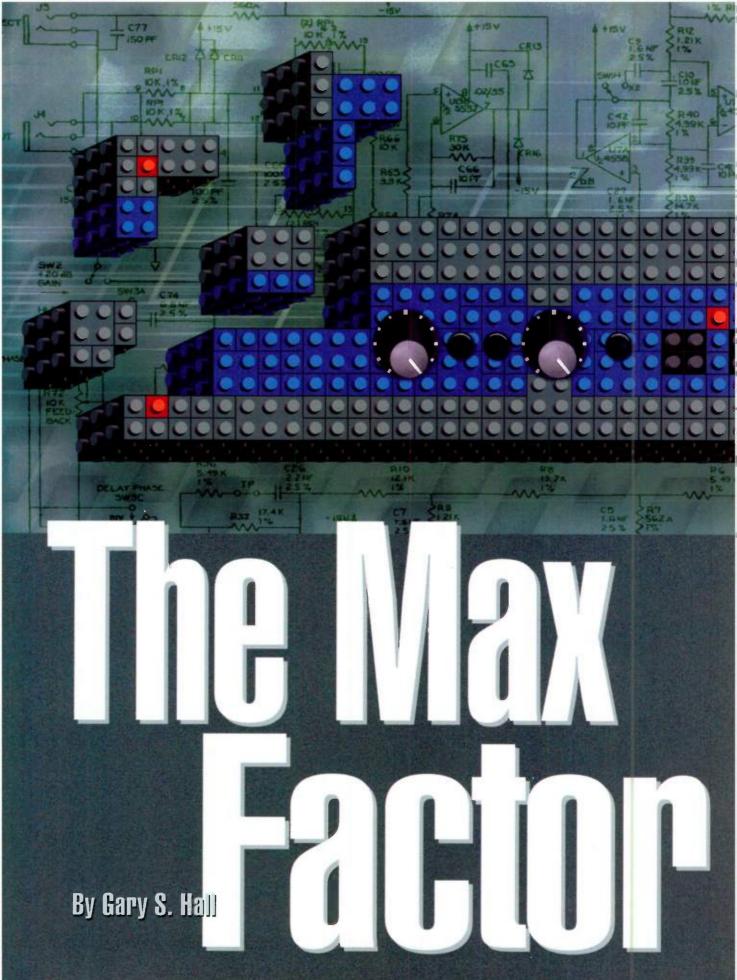
Live has support for song-length files, much like Acid Pro's Beatmapper, and it includes a Warp Points feature that lets you tie the session tempo grid to markers within a sample. Any tempo variations within the file can be followed by the rest of the session, or a file with varying tempo can be made to conform to an unwavering tempo grid. Live will also sync to MIDI Time Code or MIDI Clock, and it plays

nicely with other applications running on the same computer.

In terms of sheer features, Live is the clear winner over its Mac-platform competition. Its looping performance and ease of use are as evolved as any Windows entry—equal to Acid Pro, yet very different and wonderful in their own right. Easy to use, reliable in operation, and sexy to look at, the program competes well against all other loop sequencers. By adding a well-planned real-time layer, Live has carved out a unique and distinctive niche among loop sequencers. It's certainly worth a test drive by both Mac and Windows adherents.

Todd Souvignier *is a cofounder of Exploit Systems, Inc. (www.exploitsystems.com) and author of the* Musician's Guide to the Internet, *2nd ed. (Hal Leonard Publishing). Visit him online at http://souvignier.net.*

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

The Max Factor

By Gary S. Hall

Build on a signal-processing classic with Max/MSP.

I n the early 1980s, I had the pleasure of conceiving and designing a true classic, the Lexicon PCM42 digital delay. At that time, I was interested in using looping delays that could be beat-synchronized with drum machines and arpeggiators, and I incorporated some of my ideas into the '42. With the resources of Lexicon, I was also able to build a great-sounding delay for general use. The original '42 became nearly a studio standard; a used PCM42 fetches a higher price today than it did when it was new.

I've always taken pride in having created the PCM42. I felt, and still feel, that there's a special richness and immediacy in creating music with long delay loops with feedback, and I'm pleased to claim my little bit of history in this area.

Recently, I've been thinking about what I would do if I were to redesign the PCM42. Among the things the PCM42 would need most is the ability to dial in delay times as beats per minute and note values. There would be other changes, as well:

- My updated PCM42 would be entirely software, built with a "construction kit" such as Cycling '74's Max/MSP.
- It would be a live-performance environment, with analog inputs and outputs.
- Delay loops would be beat-synchronized, with options for master or slave operation.
- It would be semimodular, with multiple units that could be connected in series or in parallel in any combination.
- The device would offer multispeaker surround outputs as an option.

I found that I can do all of this within the Max/MSP environment, and I'm pleased with the musical results I'm getting. In this article, I'll give you a full working multichannel real-time looping device with sophisticated features, and I'll show you exactly how to build it. You can build the system from scratch or just download the completed program from the EM Web site for instant fun.

THE MAX FACTOR
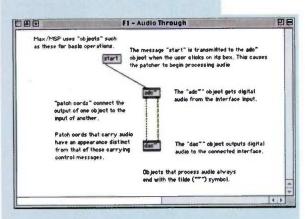
FIG. 1: Max/MSP's Audio Through Patcher accepts a stereo signal from the audio interface and passes it directly to the audio output. To understand Max and the operation of these example Patchers, pay special attention to the comments.
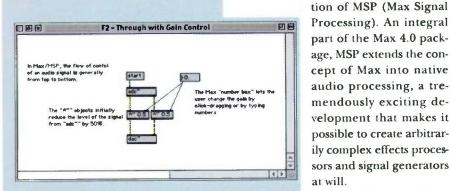
FIG. 2: The Through with Gain Control Patcher adds stereo gain control to the simple Audio Through. The initial gain is 0.5, but the gain can be controlled by click-dragging on the number box.
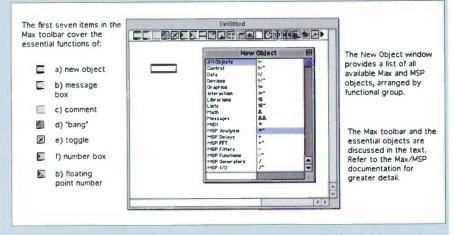
FIG. 3: When a Max/MSP Patcher is unlocked, the Object toolbar appears at the top of the window. Although there are a large number of different Max/MSP Objects available, the seven leftmost items in the toolbar are the ones that are essential to nearly any project.

THE MAX/MSP ENVIRONMENT

Max/MSP, published by Cycling '74, is a visual programming language for both music and audio. Originally developed by Miller Puckette, Max has been available since 1991. Max offers something that I've never found elsewhere: a complete software "construction set" of basic I/O, control, and processing routines that can be connected to create any function imaginable. It is truly the visual programming language of MIDI.

A couple of years ago, Max took a huge step forward with the introduction of MSP (Max Signal Processing). An integral part of the Max 4.0 package, MSP extends the concept of Max into native audio processing, a tremendously exciting development that makes it possible to create arbitrarily complex effects processors and signal generators at will.

One of the nice things that Max has always offered is its free run-time engine. With it, any Max program (called a Patcher) can be loaded and run by anyone. Controls in the user interface are fully active, but no changes can be made to the Patcher itself. Thanks to this policy, Max/MSP developers can distribute their work freely.

HARDWARE AND SOFTWARE

Max/MSP runs exclusively on the Macintosh, although a PC version is under development. OS 8.6 or higher is required (I run OS 9.1), but OS X is not yet supported.

The device we'll build is a real-time audio processor designed to work with live audio inputs, so a stereo or (preferably) multichannel audio interface is needed. Max/MSP supports ASIO and Apple Sound Manager drivers. I've used the program successfully with an Emagic EMI 2/6 and a Mark of the Unicorn 828.

Because it is based on long delay loops, the Patcher we're going to create is memory hungry. It doesn't require much for essential operation, but the more delays you use and the longer they are, the more memory you will need. As a benchmark, if you run at a 44.1 kHz sampling rate, eight seconds of delay memory (equivalent to four bars at 120 bpm) consumes about 5.6 MB.

To run the Patcher, you need some form of Max/MSP. Readers who already own Max/MSP can jump right in, of course, but if you don't yet have the program, you can download the free Max run-time engine or the complete Max package from Cycling '74's Web site (www.cycling74.com).

Running the virtual delay unit is fun and (I hope) musical, but what I really want to do in this article is show you how to modify and customize the program to meet your own needs, as well as to create your own processing and synthesis programs from scratch. This is the excitement of Max/MSP, and you should definitely give it a try.

Luckily, Cycling '74 allows you to download the complete program for a free 30-day trial. That's easily enough time for you to work through this example and some of the excellent tutorial material that is included with the download. Within your 30 days, you can

THE MAX FACTOR

FIG. 4: The Help window for any Max or MSP Object is a completely commentated working Patcher that illustrates the full function and options for that Object.
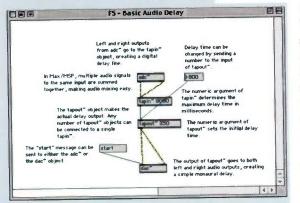
FIG. 5: This Patcher is a simple audio delay line with variable delay time. Left and right audio inputs are mixed together, delayed, and then distributed to both audio outputs.
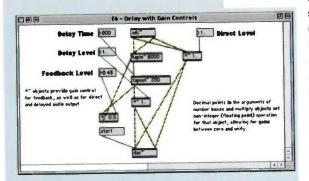
FIG. 6: This version of the delay line adds controls for feedback gain as well as level controls for direct and delayed audio. The decimal points in the arguments for the audio multiply (*~) Objects and in the number boxes that set the gains are necessary to make those Objects work in fractional values, in this case zero to unity gain.

do anything you want; after that the program will refuse to operate until you get an authorization code from the manufacturer. It's a good and fair system. I offer one caution, though. If you're running the trial version, *do not* try to change your system date and time. That will cause the program to immediately time out.

The Max/MSP Patchers shown in this article are available for download as a single archive from the **EM** Web site. To help in coordination, Patchers are identified according to the figure in which they appear. To get the most from this article, get Max/MSP (demo or full version) and build these Patchers yourself in addition to studying the downloadable versions. If you like, however, you can also download the Max/MSP run-time environment and use any or all of these Patchers as they are.

MAX/MSP AUDIO PROGRAMMING

In an article such as this, there's no way to describe the complete ins and outs of programming with Max and MSP. For that, the best source is the excellent documentation, tutorials, and Help system that come with the package (also see the review of Max 4.0/MSP 2.0 in the April 2002 issue). But let's take a look at the basic principles of operation.

In Max, low-level functional blocks, which are called Objects, are connected together in Patchers. Fig. 1 shows a simple Patcher that passes stereo audio from input to output. Notice the descriptive text that appears alongside

the Patcher itself. Comments such as those can be instructive to someone attempting to use your Patchers or to you to remind you how a Patcher works. You can download and run this Patcher, but the best way to learn is to build it from scratch in Max/MSP.

There are three kinds of Objects used here. The "adc~" Object gets input from the audio interface, "stereo by default." Audio data is transferred from the outputs of adc~ to the inputs of the "dac~" Object by virtual patch cords. The dac~ Object outputs stereo audio to the audio interface.
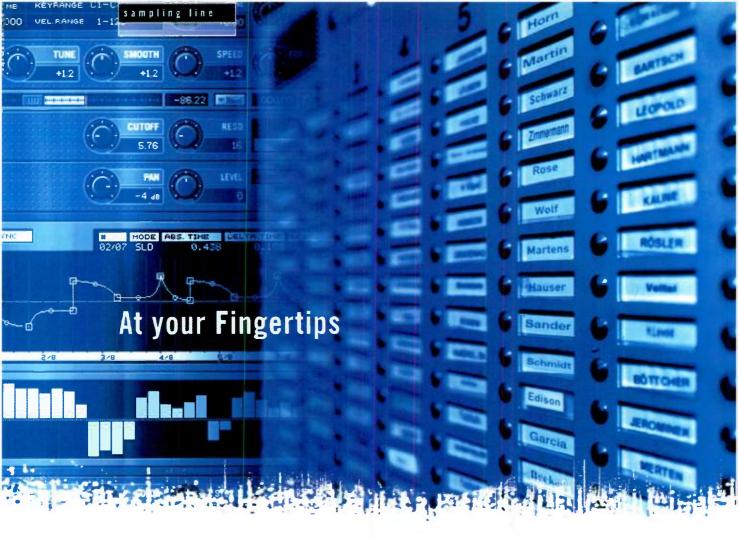
The remaining Object in this Patcher is a message box that contains the word *start*. A start message is required to initiate processing of audio. To make the Patcher operate, click on the start message. A separate stop message can be used to halt processing.

There's an important distinction in Max/MSP between Objects that handle audio data and those that deal with control and MIDI information. Max Objects for control and MIDI process data only when stimulated by the user interface or incoming messages. MSP audio Objects, on the other hand, process audio data continuously.

MSP audio-processing Objects are distinguished by a tilde ("~") sign after their names. In many cases, Objects of the same name are available in control and audio form. For example, the Object "*" will multiply two numbers, whereas the Object "*~" multiplies numbers in a continuous stream of audio data for gain control and modulation, as shown in **Fig. 2**.

Note that Max supports the Apple Sound Manager and ASIO drivers. To use your ASIO interface, put the driver into the ASIO Drivers folder in the Max application folder. Then, open the DSP Status window by double-clicking on an adc~ or dac~ Object in any Patcher (or by selecting DSP Status in the Options menu) and make the appropriate selections for your interface.

To build and modify Patchers in Max, you unlock the Patcher window by clicking on the lock icon next to the Close box. If you're running the Patchers with the free Max run-time application,

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The range of the number box can be set here in the number Inspector. By default, the initial range is unlimited, and goes to negative as well as positive infinity.
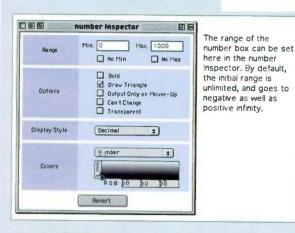
FIG. 7: The Number Inspector (select the number box and hit Command + I on keyboard) defines important properties of the integer or floating-point number box, including the number range. Defining the range of each box appropriately will help to make your audio patches behave as you intended.

The core functions of the audio delay have been incorporated into a "patcher" object. This simplifies programming and helps to maintain good visual organization.

By using "segmented" patch cords, patchers and their sub-objects can be kept orderly and easy to read.

FIG. 8: This figure shows the audio delay-line function encapsulated in a Max Patcher Object. Double-clicking on the Object labeled "patcher delayline" will open the second window, revealing the contents of that Patcher Object. Inlet and outlet Objects are used to designate inputs and outputs to appear on the completed Patcher Object.

that icon doesn't appear. Try creating a new, empty Patcher. With the Patcher window unlocked, you will find that it is easy to get started by dragging Objects from the upper toolbar and then connecting their outlets (which are found along the bottom of each Object) to the inlets of other Objects. When you drag from the Object tool, which is leftmost in the toolbar, into the main Patcher window, a handy index of the available Objects pops up, and it is sorted into functional groups (see **Fig. 3**).

Max/MSP uses a Help system that I really like. When a Patcher window is unlocked, you can pick any Object and then go to the Help menu and get a Help screen specific to that Object. The Object Help window is itself an active Max Patcher that illustrates the full function and options

for that Object, complete with comments. You can run the Help Patcher Object as is, or you can unlock it and modify it to see how the Object behaves. **Fig. 4** shows the Max Help window for the adc~ Object.

BASIC AUDIO DELAY

Now we can have some fun. In Max/MSP an audio delay line requires two separate Objects, which are "tapin~" and "tapout~." Tapin~ creates the delay unit and uses a numeric argument that defines the maximum delay in milliseconds. You then connect the output of the tapin~ to tapout~'s input. Tapout~ has a numeric argument that defines the actual audio delay; that can be changed by sending a number to the Object's input.

In the Patcher shown in **Fig. 5**, the left and right audio inputs are connected to the input of tapin~. In Max/MSP, audio sources connected to the same input are summed together at unity gain, which makes audio mixing easy.

If you run the Basic Audio Delay Patcher, you'll find that it does in fact delay audio. However, audio delay is not much fun unless you have a way to mix it with the direct signal and to feed the delayed signal back to the input with some attenuation. **Fig. 6** shows an audio delay that provides those functions.

At this point, a couple of notes about numbers in Max/MSP are in order. There are two kinds of numbers available: integer numbers (0, 1, 2, and so on) and floating-point numbers (floats) that can have noninteger values (0.5, for example). By default, Max uses integers. Audio gains, though, are usually fractional (in a range of 0 to 1.0, typically), so it's important to use Object types that are compatible.

Arithmetic Objects such as the *~ Object are made compatible with floating-point numbers by entering an initial argument with a decimal point. Similarly, number boxes such as those we're using to vary the gain values come in two types. These are represented separately in Max's toolbar and are denoted by a decimal point in the floating-point number box.

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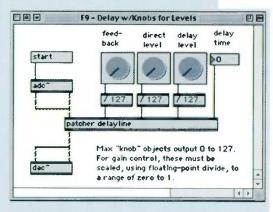
FIG. 9: The knob Object can be used to set gains instead of number boxes. Knob Objects in Max output numbers ranging from 0 to 127. To use these to set gain at a maximum value of unity, it is necessary to divide the raw output by 127, using a floating-point divide, as indicated by the decimal point in each divide Object.

FIG. 10: The Hide on Lock command in the Object menu is a powerful means of maintaining Patchers so that they look clean and neat to the end user, with wiring and Objects that the user doesn't need to see hidden from view.
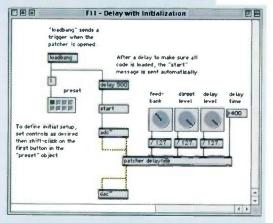
FIG. 11: The loadbang Object triggers specific actions when the Patcher is opened, ensuring that the Patcher always comes up in a known, useful state.

It's also important to limit audio gains to appropriate values, usually unity. To limit the range of values in a number box, unlock the Patcher, select the number box in question, and hit Command + I on the Mac keyboard. This opens a dialog that controls the numeric range, as well as other properties of the number box (see **Fig. 7**).

By the way, if your audio interface provides a direct-monitoring path, you can delete the portion of the Patcher that feeds the direct signal to the output. That is a better way to monitor the direct sound, because routing audio through the computer always entails at least a few milliseconds of latency.

KEEP IT CLEAN

I will now reveal to you the secret of making useful programs with Max/MSP: make sure you keep your Patchers neat. That's the difference between a hacked-up Patcher that falls apart and a complete, usable, and maintainable program. Keep your Patchers organized, and you can do anything you want.

Max provides excellent facilities to build complex functions with good organization, visual sense, and flexible design of the user interface. To

start with, you can encapsulate an entire Patcher in a single Object. When you make a new Patcher Object, a subwindow automatically opens where you can construct an entire function and link it into a larger Patcher.

Fig. 8 shows our delay module with the core delay function incorporated into such a subpatch. Inputs and outputs are defined in the subpatch by using "inlet" and "outlet" Objects, which appear like the inputs and outputs of a native Max Object. When configuring a subpatch in this way, it's a very good idea to add labels for the inlets and outlets. Number boxes inside the Subpatcher are handy for debugging, as well.

I also cleaned up our patches by using right-angle lines for patch cords instead of diagonals. To do that, select the item Segmented Patch Cords in the Options menu. When connecting Objects, you can create breaks by clicking on the cords with the mouse. I use segmented patch cords in all of my work with Max, restricting use of diagonal cords to short runs where segmented cords would be awkward.

It also helps to organize Objects that connect to each other into vertical columns or horizontal rows using the Align Objects command. Simply select all of the Objects to be aligned and use the keyboard shortcut (Command + Y) to snap the selected Objects into line.

Number boxes are also not the ideal way to control audio gain. Max provides slider and knob Objects for more conventional audio control. **Fig. 9** shows our patch with knobs instead of number boxes for the gain parameters. The knob Object outputs numbers from 0 to 127, but the gain-control multiplier needs to see a range of 0 to 1.0. To convert, divide the knob output by a fixed value of 127 using a floating-point divide so decimal values between 0 and 1.0 will be output.

Another way to help maintain visual organization is to hide connections and Objects that the end user doesn't need to see. To do that in Max, unlock the Patcher, select the item or items you wish to conceal, and then select the

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FIG. 12: This Patcher fragment converts a bpm value into a delay value corresponding to a 16th note at the designated tempo. The send Object allows this value to be conveniently routed to any number of destinations.

FIG. 13: The delayline Patcher Object must be modified to calculate a final delay value from the 16th-note value generated by the bpm master calculation and a multiple of that pulse entered by the user.
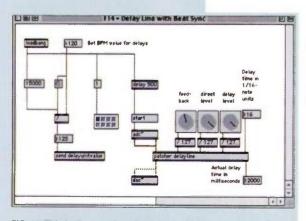
FIG. 14: This is the complete delay line with beat-sync calculation. All Objects and patch cords have been unhidden to help in tracing, and a single loadbang Object is used.

command Hide on Lock from the Object menu. When you relock the Patcher, all of the Objects that were selected will be concealed, as shown on the right of **Fig. 10**. All patch cords and Objects that the user doesn't need to see are now hidden from view.

I elected to leave the delayline Patcher Object visible for two reasons. The first is that in order to edit a Patcher Object, you have to double-click on it while the main Patcher is locked. If the Patcher Object is hidden, you can't do that. Second, it is also helpful to embed the main function of a Patcher in a single Patcher Object and then keep that Object visible. For one thing, it helps to remind me of what that particular Patcher does for a living.

Making your Patchers orderly is an art, and you should adopt the practice from the start of your work with Max. Spaghetti Patchers get out of hand very quickly.

MAKE IT NICE

It's a good idea for you to include some initialization so that your Patchers come up running in a known state. An Object called "loadbang" can help make your Patchers initialize properly. **Fig. 11** shows the previous version of the delayline Patcher with initialization added to load a known set of values and to start audio processing automatically.

In the Patcher shown in Fig. 11, I've unhidden some of the Objects and cords and added notes to make clear what's happening there. Whenever the Patcher is opened, loadbang sends out a trigger pulse called a *bang*. In this case, the bang goes to two separate places.

The first is to a Max preset Object. This is one of Max's magic functions, storing a preset for an entire Patcher window. In this illustration, a preset has been loaded into "memory 1." Loadbang triggers the message box, sending the number "1" and loading that setup.

At the same time, loadbang triggers the "delay 500" Object. After a half second, this Object sends the start message to begin audio processing. That ensures that the whole Patcher is up and running before processing begins.

ADDING BEAT SYNC

So far we've managed to create a fairly well-behaved audio delay line with an uncluttered user interface. You're probably thinking that there must be more to life than this.

There is. The good news is that because we've built up our delay line from scratch, we understand it well, and we are in a position to customize it however we like. To fulfill my original wish list, the next thing we are going to add is a way to synchronize delays with a defined tempo. The simplest way to do that is to provide a means for entering a tempo value in beats per minute (bpm) and then change the entry of delay time so that delays are expressed as a number of defined units, such as 16th notes.

Entering a tempo value is easy; it's just a number box. The trick is to convert the bpm number to a value in milliseconds per unit of time (we will use

THE MAX FACTOR

the 16th note for this). This value is then fed to the delay line, which next

FIG. 15: This version of the delay has been further simplified by incorporating I/O and control scaling into the delayline Patcher Object. This makes it easy to create additional delay sections by copying and pasting.

My updated PCM42 would be entirely software.

FIG. 16: To duplicate delay lines, unlock the Patcher, select all the Objects to be copied, and hit Command + D on the keyboard. It is usually necessary to do some initialization of parameters after duplication.

multiplies it by the defined number of delay units to get the actual delay value in milliseconds, as required by the tapout~ Object.

Fig. 12 shows a Patcher fragment that lets the user enter a tempo in bpm and converts it into a delay value corresponding to a 16th note in 4/4 time. This fragment will be incorporated into our delayline Patcher to provide simple beat synchronization. Take a moment to look over the diagram and read the notes in this example, as some things happening there may not be completely obvious.

To calculate the value in milliseconds, first divide the number of milliseconds in a minute (60 times 1,000, or 60,000) by the bpm value entered by the user. To get the 16th-note value, divide the quarter-note result by four. We can achieve the same thing in fewer steps by using 15,000 (60,000 divided by 4) as the dividend.

In this case, the Max divide ("/") Object doesn't quite work as we'd like it to, so we have to do a little bit of extra work. In the Max / Object, the left input accepts the dividend value, which in this case is fixed at 15,000. To make sure this value gets loaded on startup, a loadbang Object is connected to a message box containing the value 15,000, and this is connected to the / Object's left input.

The bpm value entered by the user becomes the divisor and is sent to the right-hand input of the / Object. However, in Max the computation of a value is triggered by the receipt of a number or a bang at the left input. We can enter values all day at the right input, but no new values for the output will be generated until we send a message to the left input. This is the function of the "button" Object, which is connected to the bpm entry box. When this Object receives any message, it generates a bang in response. The bang output is connected to the left input of the / Object, and that causes it to generate a new value whenever the user enters a new value for bpm.

If you've built this Patcher fragment from scratch, you may also have noticed that it doesn't work right away. That is because the value "15,000" is not actually transmitted until the Patcher is saved, closed, and reopened. You can get around that by locking the Patcher and clicking on the message box, which triggers it to send its message to the / Object. After that all entries of bpm will generate a new value in the number box connected to the output of the / Object.

The final output of this fragment is provided by the "send" Object. Send and "receive" Objects are among Max's great gifts to help keep your complex Patchers legible and maintainable. Send and receive Objects always have a name.

THE MAX FACTOR

Any message sent to the input of a send Object will be routed automatically to
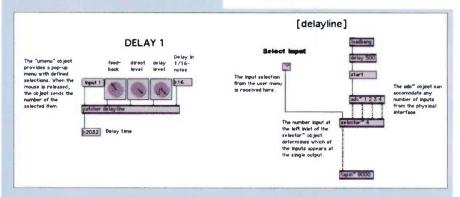
FIG. 17: The selector~ Object provides an easy way to select an audio input by a numeric value. Combined with the umenu Object, it becomes easy to control audio signal routing in a multidelay Patcher.

FIG. 18: The umenu Inspector allows you to enter a text list, delimited with commas, for the pop-up menu. When an Object is selected, umenu outputs a number corresponding to that item's position in the list, starting with 0.
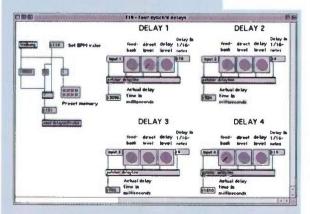
FIG. 19: You can create as many delay sections as you wish, all synced to the same bpm value. In this example, each section is fed by a different analog input, but you can also have a single source going to multiple delay sections.

any receive Object that has the same name. That makes it easy to send a single message to multiple destinations. Because we plan to create multiple delay lines tied to the same tempo, it's appropriate to use the send and receive Objects.

Now that we've derived a time value corresponding to a 16th-note pulse, we need to modify our delayline Patcher to receive this value and calculate delay times as a multiple of it. Fig. 13 shows the Max code that does that, first as a fragment and then incorporated into our delayline Patcher Object. Fig. 14 shows the complete Patcher that includes the bpm delay setting as well as specification of delays as a number of 16th-note units.

MULTIPLE MODULES

Although this seems like a lot of work just to get an audio delay line, now the real fun can begin. Thanks to the wonders of software, we can turn our single delay loop into any

number of delays, all synchronized to a common tempo.

Before doing that, we need to simplify our Patcher a little more by moving some of the functions that remain outside the delayline Patcher Object inside, as shown in Fig. 15. By moving the audio I/O and scaling for the controls into the Patcher Object, we've made our main delay section compact and self-contained.

It now becomes quite easy for us to duplicate the delay section using standard Duplicate or Copy-and-Paste commands (see Fig. 16). All you have to do is unlock the Patcher, select all the items in the Delay 1 unit, and hit Command + D. The selected items are duplicated, and you can move them as a group and change the label as needed.

When you duplicate a group of Objects, they come up without any parameter values set, so it's necessary to go to the controls and set them to something useful. You also need to reenter the bpm value so that it gets transmitted to the new, duplicate module. After doing those things, Shift-click on the first button in the preset Object we added earlier to create a valid setup that will be loaded the next time the modified Patcher is opened.

INPUT SIGNAL ROUTING

Hopefully, you're now enjoying playing your instruments through your new dual-synchronized audio-loop delay. You may be thinking that it's kind of lame, though, because both delays get the same input, and the delays come back from both speakers equally.

There are any number of ways to control signal routing in Max, including drawing patch cords from point to point. For Patchers such as the one we created, I prefer to use pop-up menus to choose an input for each delay section. Fig. 17 shows our first delay unit modified to let the user choose from a list of inputs, along with a fragment that shows the changes needed with the delayline Patcher Object to accommodate menu selection. (Patcher Objects appear in brackets in the title bar of the edit window to indicate that you're

THE MAX FACTOR

editing a Subpatcher within a larger patch rather than a standalone Patcher.)

FIG. 20: A simple left-right pan can be built using a knob control and simple arithmetic to create complementary control signals for each channel.

FIG. 21: Four-channel panning is a little more complex because left-right and front-back controls must be created separately.

The user menu ("umenu") Object provides a pop-up list that is defined in the Object's Inspector dialog (accessed by selecting the Object and then pressing Command + I), as shown in **Fig. 18**. Items in the list are delimited by commas. When the user clicks on the umenu Object, this list of items pops up. When an item is selected,

umenu outputs a number corresponding to the item's position in the list, starting with 0.

Within the delayline Object, an audio "selector~" Object controls the actual audio source routed to tapin~. The selector~ provides some number of inputs, defined by the number that follows the name, and one input for control. The number routed from the user menu goes to the left-hand input of the selector~. Selector~ routes one signal at a time to its output, depending on the number sent to the left-hand control input. An input of 0 switches all signals off.

Up to this point, we've used the adc~ Object in its default Stereo mode. But this Object can accommodate any number of inputs from a physical interface; simply list the inputs in the Object's name. To illustrate, adc~ is set up in **Fig. 18** to accommodate four inputs. It could just as easily be 8 or 16 ins.

You can also route the output of one delay unit to another's input by using send~ and receive~ Objects, which are the audio equivalents of the send-receive pair that we used for the master tempo. Space precludes detailing that in this article.

Now that you have a way to set different inputs for each delay unit, you might want to create even more iterations of the core delay, as shown in **Fig. 19**. Go ahead and have some fun with it.

PANNING AND MIXING

The last step we're going to take with our multisynced delay Patcher is to add output panning, first in stereo and then (for those with the means and inclination) in quad. For simplicity's sake, we'll stick with straight linear panning. Cycling '74 provides some excellent examples of equal-power panners and other variations in the user documents, and these are well worth studying and implementing. For the purpose of this exercise, though, we'll keep it simple.

Fig. 20 shows our main delay section with a stereo pan pot added, along with the portions of the delayline Patcher Object that are changed to accommodate

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MSP extends the concept of Max into native audio processing.

it. The key here is the creation of two gain control signals from the single

FIG. 22: With the full version of the multidelay Patcher, you can route and control as many delays as you like, with full 4-channel panning and common tempo sync.

pan-pot output, accomplished by subtracting the maximum control range (127) and then multiplying by −1 to invert. After that both signals are scaled and sent to a pair of audio multipliers that implement the actual pan.

Our last trick will be to do the same in quad. Max provides a convenient *x-y* control that makes doing that easy. To make the actual panning work, we first have to create the left and right pan outputs and then use four additional audio multiplies to implement front-rear panning separately for left and right (see **Fig. 21**).

Fig. 22 shows the final result (well, *a final result*) of our efforts: a group of four long audio delays that are coordinated precisely with a common tempo. Try this Patcher out with slow-attacking

guitar, synths, vocals, and so forth. You are guaranteed a spacious experience, especially if you can listen in quad! Of course, there's no reason to limit yourself to four modules. You can add as many as you like, until processing power or memory runs out.

MAKING IT YOUR OWN

We've come a fair distance with our quad multibeat-synced delay lines, but the real purpose of this article has been to get you comfortable with creating and modifying audio-processing Patchers in Max/MSP. You should now be able to build your own creative signal-processing tools. Even starting with the Patcher presented here, you have ample room for expansion. Here are a few ideas if you want to continue exploring on your own.

- Add a function to accept a tapped tempo to define bpm.
- Add a MIDI-clock output to drive external sequencers, drum machines, and the like at the designated tempo.
- Add a MIDI-clock input. (Take note: this is actually harder than it sounds!)
- Modify the signal routing so one delay unit can feed into another.
- Add MIDI-control inputs so all parameters can be changed with a MIDI control surface.
- Add audio recording so you can grab your jams on the fly.
- Add time-based automation so all control moves can be recorded.

All of this is possible within Max and MSP, although some tricks are more difficult than others. The real excitement will be in generating and implementing your own concepts. With Max/MSP there's simply no need to be limited by what's available in current fixed-function products, because you can build anything you like, on the spot, and totally from scratch if need be!

Gary S. Hall *consults, writes, and composes from his home base in Alameda, California, and from custom remote facilities in Bahia and the Utah outback.*

The Universe for a Song

You don't need a big-budget studio to put "space" in your DVD.

By Tim Tully

I've been producing a commercial DVD titled The Universe, and one of the most challenging aspects of the project has been building the 5.1-surround soundtrack.

The primary content of The Universe is a huge number of animated stills and movies showing the amazing astronomical images being sent to Earth by various space-based telescopes. The well-known Hubble Space Telescope captures images of cosmic objects ranging from within our own solar system out to unbelievably distant extragalactic regions. The less famous TRACE and SOHO telescopes shoot incredible close-up videos of the surface and corona of the Sun.

From the initial stages, I planned to create the entire project in my home project studio. The tools were readily available, and I believed I could push the project through—at least to the replication stage—with only moderate additions to my audio-video setup. I knew it would be a challenge but that it would ultimately be worthwhile for a number of reasons.

I cut the video, including the music and narration, in the Windows-based Vegas Video from Sonic Foundry. I use Vegas because its interface is remarkably intuitive and fast, and its real-time preview capability saves me huge amounts of time. My plan was to encode the edited video from Vegas to MPEG-2 and then move it to a Macintosh G4/867 MHz to author the DVD with Apple's affordable DVD Studio Pro software.

THE CHALLENGE

I knew that to be competitive the DVD would have to make the most of its surround-sound capabilities. I also knew that it would take some work. The first

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hurdle I encountered was that most of the music I was using, composed by Mix magazine's Paul D. Lehrman, was mastered in stereo. In concept, converting Lehrman's stereo mix into surround was simple. I wanted to avoid any aural trickery, such as extracting musical parts and flying them through figure eights behind the listener's head. Instead, I decided that the most effective approach would be to sweeten and process the music in ways that took subtle advantage of the surround channels.

The first and simplest step would be to put reverb and possibly some delay in the rear speakers. That would strongly establish the viewer's sense of being in the same space as the images. As you may know, there is no sound in outer space. Yet this literal reality has been violated in so many sci-fi TV shows and movies that an entire vocabulary of outer-space sounds has become common parlance. For example, big nearby ships emit a loud, deep hum; lasers make a quick, resonant filter sweep; passing spacecraft and various meteorites go whoosh; and so on.

Those conventions are so well established that people not only accept them but expect them. That's fine with me,

because I can use that vocabulary to enhance the audio experience of my DVD. My goal, therefore, was to envelope the viewer in a sizable environment with a big, smooth ambience that would suggest the feeling of being in an immense, intergalactic space.

The next step would be to add percussion tracks, principally to the sides and rear. Adding melodic and harmonic sweetening would also be possible, but much more difficult without sheet music or even chord charts to go by.

THE GEAR

To turn my stereo studio into a surround room I had to get some new tools, including Mark of the Unicorn's Digital Performer 3 sequencer and 828 digital-audio interface to run on my Mac G4.

Digital Performer 3 boasts serious surround capabilities: the program has several types of panners, each with different strengths and functions, and it even lets you mix with a joystick. The panning functions were especially important because I use an analog mixing board, which I've been reluctant to part with for various reasons (good busing, lots of aux sends, onboard EQ, and it's quiet). I also needed to use several MIDI instruments to get the sweetening and sound effects I wanted.

The new 828 seemed like an ideal solution. It uses FireWire (rather than the slower USB) to communicate with the computer. FireWire has been a blessing when I have worked with the greater bandwidth demands of video. The 828 also has eight analog inputs and outputs—more than I would need—and it offers excellent software control and compatibility with Digital Performer 3.

For speakers I picked a 5.1 set
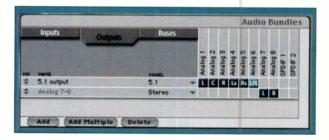
FIG. 2: Digital Performer 3's intuitive Audio Bundles window lets you move tiles around to match the I/O of the software with the 828 hardware interface.

from Cerwin-Vega. They're consumer speakers, but they are accurate in addition to being compact and affordable. A more difficult task than choosing the speakers was finding a way to feed them the signals.

If you're turning a stereo studio into a surround room, one of the things you absolutely need is an amplifier that supports surround sound. On the output side, that's not too difficult. For as little as $175, you can find an AV receiver with five outputs for your left-front, right-front, and center-front speakers, as well as your left-surround and right-surround (rear) speakers. You also typically get a line-level output for a powered subwoofer.

Somewhat more difficult and expensive is finding an amplifier with six discrete line-level inputs. Some AV receivers have them, and some don't. Project-studio budgets usually dictate that you go for consumer gear. Because you can't count on help from the salespeople at consumer stores, you have to do some research to find the right box. (The Web is invaluable.) Check the manual and, more importantly, the actual rear panel to make sure the unit has six analog inputs (usually RCA jacks) labeled something like DVD Six Channel or SDDS Inputs.

In the end I chose a Kenwood VR-407. Because it's about a year old, I was able to get it for $185. It's rated at 100W per channel, has a reasonably simple setup (and an awful manual), and even says 6-CH Input on the front.

Once you install the driver for the 828, remember to open the 828 control panel and set it for a 48 kHz sampling rate (see Fig. 1). That's important

FIG. 1: The 828 control panel lets you set a number of parameters, including the all-important sampling rate.

because Apple's A.Pack, which is a good tool for encoding your final mix to MPEG-2, doesn't work with any other sampling rate. You don't want to get all the way through a project only to realize it's set to 44.1 kHz, and you have to reburn it at 48 kHz. The good news is that once you import a sound file as a Soundbite into Digital Performer 3, it converts the sampling rate simply and quickly.

THE HANDS GET DIRTY

Configuring all that gear took some time. Although Digital Performer 3 and the 828 are perfectly suited for surround mixing, their manuals focus primarily on recording in stereo to tape, and you have to dig a bit to set them up to do surround. The primary distinction between the two approaches is that the final output for DVD consists of six mixed audio files on disk and not six tracks on tape.

My job was further complicated by the need to use MIDI to sweeten the music and do sound effects. Although Digital Performer 3 does MIDI and audio, and though its surround mixing works as advertised, basic MIDI tracks themselves cannot be panned in 5.1—only in stereo. That's because MIDI synths and sound modules are stereo devices, and the MIDI Specification doesn't accommodate surround mixing.

So just how do you route, monitor, and mix a 5.1 soundtrack using MIDI and audio? The answer isn't immediately obvious, but here's the solution: you have to use the audio outputs of the 828 for monitoring, not for recording. When I first realized that, I unconsciously resisted the notion and wasted time looking for another approach. That may have been because the 828's outputs are of such high quality and are intended for final output. Quarter-inch TRS jacks with 24-bit D/A resolution seemed like overkill for monitoring. When I finally accepted that fact, though, things became clearer.

I then connected six of the 828 audio outputs to the AV receiver and hooked up the receiver's respective speaker outputs to the surround speakers. The re-

ceiver also had to be set up internally to accept six discrete line-level inputs and to route each to a separate surround speaker.

To set up surround monitoring in Digital Performer 3, you open the Audio Bundles window, where you can create various input and output configurations by moving little blue squares (tiles) around on the screen (see **Fig. 2**). For example, to set up the left-surround channel: in your Digital Performer 3 5.1 audio-output bundle, move the Left-Surround tile to the column that represents the physical 828 output connected to your left-surround speaker.

Follow that procedure for each of your six channels, and the worst is over. Double-check that the subwoofer output goes to the right place. If the Monitor amp has an output for a powered subwoofer, make sure you connect it to a powered sub. The other option you have is to send it to an unpowered sub with a passive crossover.

Assigning an audio track to a 5.1 output bundle equips the track with a surround panner in Digital Performer 3's Mixer window. That lets you send each Digital Performer 3 track to any combination of the surround speakers (see **Fig. 3**). That takes care of monitoring; the next step is setting up for MIDI.

THE MIDI QUESTION

As I mentioned earlier, I needed to compose, monitor, and mix my MIDI tracks to surround while combining them with the original music. Because MIDI tracks can't have surround panners (remember, they're triggering stereo devices), that posed a problem. Composing and recording would be simple; I could do that in mono or stereo. But how could I mix those tracks to surround? Fortunately, Digital Performer 3 offers a couple of options.

The approach I took was to bus all of the channels on my mixing board (the hardware, analog one) that were assigned to MIDI instruments to a single stereo output. I connected that stereo pair to a pair of 828 inputs (real, not virtual). I could have chosen any pair, because all of the unit's audio inputs were available; the rest of the audio was

FIG. 3: Once you set up a surround Audio Bundle in Digital Performer 3, each audio track is equipped with a surround panner.

coming from the Macintosh through FireWire.

That done, the first option was to create an input Audio Bundle for the 828 inputs that were receiving the synth signals. I routed the inputs to a Digital Performer 3 aux track and assigned the output of the aux track to the surround output Audio Bundle. The aux track, like Digital Performer 3 audio tracks, does have a surround panner, which lets you put your synth sounds where you want them.

All that routing is not free, though; it creates a latency that delays the synth sound. Digital Performer 3 has a control panel to adjust that, and it may not be a problem in some situations. Because I like recording in real time, however, I didn't feel comfortable working with the latency. So I looked for another solution.

I solved the problem with a simple expedient. I recorded my mono and stereo MIDI tracks straight in as usual. I didn't hear them in surround, but I did hear them in time. When I had a performance I liked, I recorded the MIDI track as an audio file, and the problem was solved.

Recording the MIDI as audio is a no-brainer. Solo the MIDI track, add a new audio track, and set its input for Analog 1–2. Record-enable that track. Press Record, and you will write an audio version of the MIDI track to disk. The

MIDI and audio tracks are even in sync. I heard no phasing, canceling, or other out-of-sync artifacts. I moved the MIDI track down out of the way and moved its twin audio track up to where it was accessible, and I was rocking.

MIXING TO DISK

When all of the tracks are right, your last step is to record the whole piece to disk. Digital Performer 3 has a great Bounce to Disk feature that can create six mixed Sound Designer II files from any portion of a project that you select. Those are the files that you encode as MPEG-2 for importation into DVD Studio Pro to author and burn the DVD.

▼

The panning functions were especially important.

It's as simple as making a selection and choosing a menu item. All the audio, MIDI, and effects generated by Digital Performer 3 plug-ins are combined just the way you set up the project.
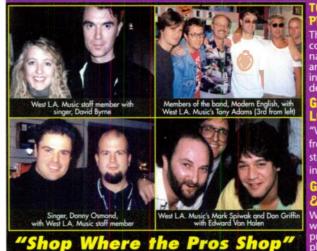
You pay a price for mutating a stereo project studio to do surround, but it's not too high. The toughest part is figuring out exactly what has to be done to get all of the players speaking the same language. Once my system was in place, I took a few small test projects from start to finish, and everything seemed to work.

At the moment, The Universe is still in production—on the video as well as the audio side. But it should be completed by the time you read this. The project offers new challenges every day, it looks and sounds great, and it presents another step forward in desktop production. If you'd like to see some samples of the project, check out www .universedvd.com.

Tim Tully is a musician, composer, and video producer who has written hundreds of articles and cowritten two books on media production.

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

SQUARE
ONE

Better Latent? Never!

Learn how to overcome latency, the demon of computer audio.

By David Battino

Imagine trying to play a duet with another musician standing two and a half football fields away. Because sound travels through air at about 1,000 feet per second, each of you would have to wait approximately three-quarters of a second (750 ms) to hear the other, making collaboration frustratingly difficult. Yet it's not uncommon for computers to impose that amount of delay on audio. For anyone trying to record music on a computer or play software

synthesizers, that delay, called *latency*, can make the process feel as though you are trying to run with a 300-pound linebacker wrapped around your legs.

In this article, I'll explain what causes latency and how to reduce it to a musically acceptable level. Because of the way computers handle audio, latency is often unavoidable. But if you understand why it happens and how to work with it, you'll be able to get the most expressive performance out of your setup.

DEFINE "LATE"

Latency simply means the time it takes a computer-music system to move audio from one place to another. Every time the signal is altered (converted from analog to digital form, for example, or passed through a reverb effect), the computer needs time to do the calculations. In addition, computers transfer audio data in chunks rather than in a continuous stream, which introduces a delay equivalent to the size of the chunk.

This chunking technique is actually used to ensure a continuous flow of audio, which may at first seem counter-intuitive. As BIAS programmer Dug Wright explains, "In the analog world, audio moves through a studio as fast as the electrons can travel, which is basically the speed of light. You can plug

FIG. 1: Optimizing your computer for audio involves finding the best balance between responsiveness (latency) and CPU usage. This control panel from Native Instruments' Reaktor, a software synthesizer, details some of the trade-offs.

in your guitar or keyboard, play a note, and its sound comes out of the speakers at essentially the same instant. In a digital system, there is a clock dividing audio into little slices (called *samples*) at the *sampling rate*. Then that little slice of audio is moved around by a network of gates that can open and close only at the frequency of another clock—about 800 MHz with a modern computer. Because the CPU has to do many things, like draw graphics and move the cursor, it divides tasks such as moving audio data into chunks or *buffers* of samples."

To see why grouping samples into buffers helps to ensure a smooth flow of audio, imagine giving yourself a shower with a squirt gun. If your finger cramped up or the gun ran dry, the stream would stop, interrupting your shower and leaving you cold. Now imagine using a bucket instead of the gun. By filling the bucket and then punching a hole in the bottom, you'd guarantee yourself an uninterrupted shower. And as long as you could add water to the bucket faster than it drained, your shower could continue indefinitely. The drawback is that you wouldn't be able to begin your shower immediately because it would take time to fill the bucket. Moreover, if you wanted to change the water temperature, you'd have to wait for the most recent squirt of water to work its way down to the bottom of the bucket.

In audio terms, the bucket would be the buffer, and the time it took the new

water to descend to the bottom would be the latency. If the buffer "runs dry," you will likely hear clicks and dropouts. Most professional audio interfaces allow you to set the size of the buffer in software (see **Fig. 1**); some let you set the number of buffers as well. Although smaller buffers reduce latency, they also require more CPU time. Every time a buffer fills up, it sends an *interrupt* message to the computer to indicate that the samples are ready to be processed. That forces the computer to process interrupts instead of audio data, which reduces the number (and complexity) of effects and synthesizer voices it can produce.

Play it safe by using a larger buffer, and you'll notice a correspondingly longer wait before hearing the effects of any musical changes you make—twisting an onscreen mixer knob, for example. Balancing the risk of glitches with system responsiveness is the art of optimizing latency; I'll cover that process in a moment.

LATENCY, STEP BY STEP
As mentioned, the total latency in a system is the sum of several smaller latencies. To demonstrate how these delays add up, Wright offers the following map, which traces the audio signal through the computer from input to output:

1. The audio interface's analog-to-digital (A/D) converter samples the audio, which usually induces a delay of about 1.5 ms.

2. The interface stores 64 to 4,096 or more samples in a buffer; it then tells

TOO LITTLE, TOO LATENT
When you're confronted with arcane control panels for juggling buffer size and CPU overhead, it's easy to forget that the ultimate goal is to make music. While most experts I spoke with described complex scenarios involving oscilloscopes, auxiliary computers, and other lab-coat apparatus, Clint Ward of Emagic shared this uniquely musical way to optimize any new audio device or driver:

1. Install the driver and start your test with its default buffer settings.
2. Launch a virtual instrument such as the Emagic EXS24 software sampler and load up a drum kit. (A swelling string sound will not work for this test.)
3. Play the drum kit in real time and make a judgment on the feel.
4. Change the driver's buffer settings to the lowest possible value.
5. Play the drum kit again and make a judgment on the feel and the stability of the driver at that setting.
6. Record track after track until the driver or the CPU falls down.

Ward remarks, "In this test you will either experience problems immediately (at the lowest latency settings) or you will eventually hit a computational wall. It will shed immense light on the maturity of the driver, the audio device, and [whether] you need a new CPU. If the driver doesn't fall down until you reach the computational roadblock, you have a potential winner. Increase the buffer size until you perceive latency, then ease off one notch. I feel you give up way too much CPU at the lowest setting when a less greedy one will still feel fine."

As Brian Smithers noted in "Desktop Musician: Musical Protocols," in the December 2001 issue of **EM**, another trick is to reduce the size of the buffer while recording soft-synth parts, then restore it to a safe level during mixdown. That technique provides fast response when you need it most—during live performance.

FIG. 2: A growing number of computer audio interfaces, such as the Emagic EMI 2/6, offer zero-latency monitoring. By routing incoming audio directly to the interface's output rather than looping it through the computer, this design allows you to hear yourself with no delay as you record.

the operating system—through an interrupt—that it's ready to hand them off. Consequently, the oldest sample in the buffer is delayed by the duration of the buffer. This delay is equivalent to the number of samples the buffer can hold divided by the sampling rate. For a 1,024-sample buffer at 48 kHz, that's about 21 ms. (Increasing the sampling rate would lower the latency, but there's no free lunch, because a higher rate fills the buffer more quickly, increasing the frequency of interrupts and the strain on the CPU.)

3. The system has to stop what it was doing (because it was interrupted) and get the audio data. Depending on the operating system, that takes anywhere from 500 microseconds (0.5 ms) to 250 ms. That delay is called *scheduling latency*, and while it isn't part of the latency added to the audio, it can force you to use larger buffers in order to process the audio in real time. (The buffer size must be bigger than the scheduling latency.)

4. The operating system then delivers the audio data to the music program through an intermediary piece of software called a *driver*. Now the application can process the data (record it, mix it, apply effects, and so on) and send it back to the audio card through the operating system to be output. This is where processor speed (more specifically, floating operations per second,

or FLOPS) comes into play. A faster processor will be able to perform more operations in the allotted time.

5. During the processing of the audio signal, different signal processors can induce delay. This is referred to as *algorithmic latency*. Certain types of filters can delay the signal by several samples (sometimes up to hundreds of microseconds); dynamics and intonation plug-ins are especially demanding. If an application is compensating for those delays to maintain sample accuracy between channels, then another few hundred microseconds of latency can be added.

6. The audio interface usually accepts data only in chunks that are the same size as the input buffers and at the same time that it's ready to hand off another buffer of incoming data. With the numbers we've been using, that means the samples are again delayed by 21 ms.

7. The audio card converts the audio back to analog, taking another 1.5 ms.

So to tally up, an audio card running at a 48 kHz sampling rate and set to 1,024 samples per buffer will induce approximately 1.5 ms (A/D) plus 21 ms (input) plus 0.5 ms (algorithmic) plus 21 ms (output) plus 1.5 ms (D/A), or 45.5 ms total latency from analog input to analog output.

OVERCOMING LATENCY

Fortunately, musicians don't have to keep track of all these little steps—playing 21.5 ms ahead of the beat so their overdubs will sync with previously recorded tracks, for example. "All modern driver architectures have a mechanism to report the latency to the host application," explains Emagic's Gerhard Lengeling, "so the application can simply move the recorded data position on the timeline" to compensate during overdubs.

Many newer audio interfaces

feature *zero-latency monitoring*, which routes a copy of the incoming signal directly to the interface's output (see **Fig. 2**). Because the copy you're monitoring doesn't pass through the computer, you hear it instantly, but you won't be able to hear any software-based effects processing. If you do want to record or monitor with effects, however, you can use the old-school solution: simply route the input signal through an outboard hardware mixer and effects, then monitor the mixer's output rather than the computer's.

Similarly, you can use an audio interface such as the Korg OASYS PCI (see **Fig. 3**), which has onboard digital signal processing (DSP) chips. This design puts the outboard effects on the audio card itself for instant aural gratification. The ultimate (albeit the most expensive) approach is to use a complete DSP-based recording system such as Digidesign Pro Tools. These systems minimize latency by doing all their processing on custom hardware, using the host computer solely for graphics and disk access. (You still get the 1.5 ms A/D and D/A conversion delays.)

DRIVE TO SUCCEED

There are ways to improve latency without hauling out the heavy artillery, though. One of the best is to update your drivers, the tiny pieces of code that handle communication between the music software and the audio interface. Under Windows, this generally means substituting ASIO or WDM drivers for

FIG. 3: Audio interfaces with digital signal processing (DSP) chips, such as this Korg OASYS PCI, can often apply effects to incoming audio. That allows you to hear effects while recording—without the latency you would get if the signals were processed on the computer itself.

the standard, comatose MME drivers. (Note that your interface and software must support the new driver type.) The default Sound Manager drivers on the Mac provide reasonable latency performance (about 11 ms), but Macs can also benefit from an ASIO upgrade. Most audio interfaces support a variety of drivers; the sidebar "Too Little, Too Latent" offers tips on testing them in your setup. For background on ASIO, WDM, and other driver standards, see "Desktop Musician: Musical Protocols," in the December 2001 issue of **EM** and also online.

As mentioned, a faster computer can reduce latency by doing an equivalent amount of processing in the space of a smaller buffer. But you can ease the load on your current CPU by turning off extensions and background tasks. Some people even replace graphics cards that handle interrupts poorly, delaying the computer's response to audio interrupts.

Upgrading your operating system

may help, too. "Possibly OS X will be *the* audio platform for its good real-time behavior," says Adam Castillo of interface manufacturer M-Audio, which has achieved latencies of less than 1.5 ms on the new Mac OS. The company has also had success under Windows 2000 and XP, clocking 1.5 ms latencies using its Delta cards, Cakewalk Sonar, and WDM drivers. A recent study at the Peabody Institute found that the Linux operating system outperformed both OS X and Windows in some configurations. You can read the study at http://mambo.peabody.jhu.edu/~karlmac/publications/latency-icmc2001.pdf.

Emagic's Clint Ward puts it succinctly: "Latency *really* occurs when someone is trying to use the native audio system on a slow CPU with no RAM and an old audio I/O device with terrible drivers."

WORKING LATENT

Whoever said "It's never too latent" probably wasn't a musician. But al-

though some amount of latency is endemic to computer music systems, it's not necessarily the show-stopper that it's often made out to be. Every day, live bands crank out killer performances while their members are standing 15 feet or more apart. Although we don't normally think of it in such terms, this separation creates a "latency" of 15 ms, which is ten times the amount easily obtainable with today's sound cards and software. When you consider that early computer musicians had to wait hours to hear a single note, we've got it pretty easy.

David Battino *is the editor of* **EM**'s *2002* Desktop Music Production Guide. *He would like to thank Adam Castillo of M-Audio, Greg Ondo of Steinberg, Dug Wright of BIAS, and Clint Ward and Gerhard Lengeling of Emagic for their contributions to this article.*

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

Shake Your Moneymaker

Take your studio from money pit to moneymaking machine.

By Jeffrey P. Fisher

Take a look around your project studio at all the equipment you have accumulated. Are you tired of spending money on gear and ready to make money with it? It's time to make a plan to boost your studio profits.

Almost every day for the more than 15 years I've owned my project studio, I've tried to uncover better ways to make more money from my gear investment. Ten simple methods have worked con-sistently for me. Perhaps they'll help you put more cash in your pocket.

RAISE YOUR RATES

Just as most employees have annual salary reviews, you should step back and look at what you're charging, too. At one point, I went 26 months without a fee increase. What was I thinking? Now I increase my rates at least 5 percent each year.

Knowing the going rate for the services you provide is essential. What are your peers and your competition charging for similar work? In the survey I conducted, rates ranged from an insulting $15 per hour to a more respectable $75 per hour. In addition, assess your experience. If you're young and green, charge about 80 percent of the average rate you dis-cover; if you possess impressive creden-tials, charge on the higher end—120 percent or more of the average.

Another gauge to use to establish your rate is whether clients complain when you increase your rates. If they do, decrease your fee a little and you've found the highest rate your market will bear. Notice I said "clients" complain, plural. One complaint from a notori-ously cheap patron should be ignored. Only when protests come en masse should you yield.

CUT YOUR EXPENSES

The easiest way for you to increase your income is to put a larger gap between what you earn and what you spend. The obvious remedy is to keep your gear lust at bay. I know it's hard. The pages of **EM** are full of stuff that you just have to have. While I'm not against investing in new toys, it helps your bottom line when you invest wisely. Take as much care purchasing new gear as you would putting your hard-earned savings into any traditional investment.

My philosophy has always been to start with a basic equipment arsenal and put an action plan together for acquiring additional gear in the future. Set aside 5 percent of what you make from your project studio as your rainy-day gear-splurging fund. Objectively evaluate your needs, industry trends that affect your business mix, any specific demands by clients, and, most of all, the profitability of any new equipment acquisition. Will you recoup its cost through increased billings?

Also, look for ways to reduce your fixed business expenses and keep careful watch over other expenditures. Rent, utilities, telephone, and Internet charges can easily get out of hand. Thankfully, they are easy to reduce, too. If you're unsure of where your money goes, record your spending habits for a few months and find ways to cut the chaff. Don't limit this technique to business expenses, either. You can target and cut many frivolous personal expenses at the same time.

Don't forget about the tax impact of business purchases. If your project studio is a legitimate business (see the sidebar, "Mind Your Own Business"), you reduce your taxable income through prudent business expenses. Therefore, you pay fewer taxes. For example, say you are in the 27 percent tax bracket, pay the 15.3 percent self-employment tax, and work in a state with a 3 percent income tax. Buying $1,000 in gear saves you about $453 you would otherwise pay on April 15. (The actual amount is $441.30, thanks to wacky IRS math.) The $1,000 business-gear purchase reduces $1,000 of earned business income to zero.

CHARGE FOR YOUR SERVICES

The majority of project-studio owners charge for the studio itself and throw in their services for free. Consider the opposite approach. Start charging significantly more for your professional experience and, instead, give the studio time away. People will pay more for a professional producer and engineer than for a room in your basement.

My clients rarely visit my studio. I do my music and sound magic and deliver the finished work to them. My fee is considerably higher than what I would earn from my humble computer-based project studio alone. I'd be lucky to get $35 per hour for the room, whereas I have no trouble billing triple that myself.

Voice artist Harlan Hogan echoes this strategy. "Talent union scale for a non-broadcast session is $333 for an hour plus 14 percent for the Health and Retirement fund," he says. "I charge $550 per hour, plus H&R, and throw in my project-studio time for free. Instead of charging for the room, I add value to my voice-over work and get paid better anyway. Plus, I have no travel time, which allows me to do more sessions." He delivers spots through ISDN, through e-mail, or by burning a CD and mailing it.

REWARD LOYALTY

Be good to your anchor clients. Don't neglect the 20 percent of your established client base that provides 80 percent of your revenue. Treat the clients well, and they'll reward you with their continued patronage and loyalty. Simply picking up the phone and inquiring about new projects or pitching my own ideas invariably results in an assignment. Sharing case studies of other successes is another way to encourage your best spenders.

Introducing a new product, service, or piece of gear? Make sure that your best clients know about it first. Also, offer them exclusive discounts or other incentives. Anchor clients are good sources for referrals to other prospects who may need your music and sound services. Don't be shy about asking for their help finding new business.

FIND BETTER CLIENTS

Stop wasting your time on nickel-and-dime gigs. It's worth the extra effort to find clients with bigger budgets. Sure, you can do band demos for $35 per hour, but I can make $600 in half a day selling on-hold messages to business clients (see "Recording Musician: Money on Hold" in the February 2002 issue).

Ad agencies, video and multimedia production companies, and general business clients usually have bigger budgets for music- and sound-production work. Find these deep pockets and pitch your most persuasive argument. A few projects from better clients can keep you busy for weeks and can pay handsomely.

DRUM UP NEW BUSINESS

If you don't have any work lined up, don't wait for the phone to ring. It's time to get up, get out, and get going to win new business. Unless you have an impeccable reputation, don't expect people to contact you. You need to initiate proper promotions that bring a steady stream of new business to your room.

Call previous clients and see if they have any new projects coming up. Make some cold calls to possible prospects, too. Write a news release about your latest accomplishments and send it to the media that reaches your target market. Put together a simple postcard about your services and send it to your contact list. Have a sale and offer a discount of some kind. Look for networking opportunities. Go where the work is. Hang out with the people who need the project-studio services you offer. Keep your name and the benefits you offer in front of buyers regularly. It's the essential way to secure your success, now and for the future.

RESURRECT OLD LEADS

Did somebody inquire about your services but never buy? It's time to check in with that person again. People move around a lot. Your contact at one company may have left his or her job, and the new employee filling that position may never have heard of you. Simply

MIND YOUR OWN BUSINESS

It makes sound financial sense to run your project studio as a legal business. Following are some basic steps to get you started. For more detailed information, see "Working Musician: Going Legit," in the February 2002 **EM**.

• Set up your business by choosing its legal structure (sole proprietorship, partnership, corporation, and so on). Consult with a tax adviser for details about the financial aspects of each business type. Contact a legal adviser for answers to liability issues.

• Make sure you meet any regulations for running a business in your town. For example, you may have to get a business license from your local clerk's office.

• File a doing-business-as (dba) with your local government if you call your business something other than your legal name. You may need a separate tax ID for your business, and some states require a sales-tax ID number.

• Open a business checking account. Deposit your project-studio income into it and pay your business expenses using checks drawn on it. Also, use a credit card only for business purchases and pay it off on time from your business checking account.

• Use bookkeeping software to track your business income and expenses. This will facilitate preparing your taxes and monitoring your financial situation. Understand the various tax consequences of your business. You'll probably need to make quarterly tax payments in addition to your annual payment.

• Protect yourself through health and property insurance. Also, consider obtaining or adding more life, disability, and liability insurance if it makes sense for your situation.

reconnecting with lost leads can generate lucrative assignments. Also, don't lose track of those job jumpers. They may have moved on to brighter horizons, and they still may remember how you helped them along the way.

ADD PRODUCTS

Products can sometimes make up 50 percent of your revenues. If you limit yourself to just selling project-studio services, you could be cutting your income potential in half. Now is the perfect time to add products to your overall business mix. Choose only those products that complement what you sell. As a composer, I sell a buy-out music-library product to supplement the income I earn from soundtrack and jingle gigs. I know one studio owner who worked out a commission deal with a local music store to sell some gear available in the room. Many musicians finished their sessions at his studio only to leave with a new toy and a lighter wallet.

STAY CURRENT

Are clients or industry trends hinting at new services you should be offering? Listen to what people tell you. Follow the industry closely. Then, act accordingly. For instance, are you set up for 5.1? I now offer video-editing services alongside my usual audio-for-video services. Finding a distinct need and filling it is a sure way to keep cash flowing in.

OUTSOURCE

If you work alone, you have only so much time to accomplish what you need to do. Consequently, there's only so much money you can make. If you hire other people or firms to handle ancillary services, you can get more done in the same time. You'll bill out higher invoices, too. You don't have to hire employees to make this work. You can hire independent contractors and other businesses to pick up your slack.

If you're recording and mixing a CD project for an act, don't let your involvement stop with the stereo mix. Subcontract a mastering house to finish the master. Hire a graphic designer to put the CD package together. Work with a duplication company to make the finished CDs. Mark up the charges incurred from these subcontractors and bill your clients the higher amount. You can easily make 20 percent or more of the mastering, design, and duplication fees through a couple of phone calls and a little oversight. Your clients will appreciate the one-stop service, too. You'll appreciate the extra pay.

Follow these strategies, and you'll see a significant increase in the money you make from your project-studio endeavors.

Jeffrey P. Fisher *is the author of* Profiting from Your Music and Sound Project Studio *(Allworth Press). Visit his Web site,* www.jeffreypfisher.com, *for more tips on moneymaking.*

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

REVIEWS

DIGITECH

GENESIS3 GENETX

A compact guitar processor aspires to more than the everyday.

By Myles Boisen

The old art-school adage "Less is more" has had its day, but it's clearly out-of-date when the topic of guitar processors comes up. For today's guitar slinger and home recordist, more is more; the more amplifier types, cabinet configurations, and effects, the better, especially when a virtual truckload of vintage amps can be packed into a device the size of a paperback book. DigiTech's contribution to the maximalist trend is the Genesis3 GeNetX guitar processor, a cool blue studio tool packed with enough power to keep plectrists drooling for months.

IN THE BEGINNING

The Genesis3 case is pleasantly sculpted into a rounded rectangle 10.5 inches wide by 6.5 inches high (see Fig. 1). Although the housing and knobs are made of plastic, it seems rugged enough to take at least a few bounces or kicks. Some of the knobs are a bit too close together for my comfort, but they're all easy to grasp and large enough for fat-fingered fretters.

FIG. 1: The DigiTech Genesis3 combines credible amplifier and speaker modeling, acoustic-guitar modeling, and effects processing with an easy-to-read display and intuitive controls.

www.emusician.com

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ksp 8

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416-492-9899
Fax: 416-492-9299

FIG. 2: All I/O connections are located on the Genesis3's rear panel, including a standard guitar jack and an ⅛-inch stereo minijack for audio input. The MIDI In port doubles as a connection for the optional Control X foot controller.

Input and output connections are on the rear panel, just above the front control panel when viewed from overhead (see Fig. 2). Because the jacks are out of sight during normal use, DigiTech has thoughtfully indicated their functions and locations on the front panel. Both analog audio outputs are unbalanced ¼-inch TS jacks. The Digital Out is co-axial S/PDIF, and the CD in and head-phone out are ⅛-inch TRS minijacks. The power cable uses a standard un-grounded AC connector with a 9V in-line transformer instead of a wall wart.

SUPER MODEL

The Genesis3's front panel is organized around three horizontal groups of knobs. The bottom row of five pots should be familiar to any guitarist, because it duplicates common amplifier features: Gain, Bass, Mid, Treble, and Amp Level. As on most amps, the initial Gain knob controls the preamp; you can turn it down for clean tones or crank it up for overdrive effects. Amp Level governs the output of the emu-lated amp combinations, with a final output stage control located on the back panel. A numerical readout (0 to 99 for gain and –12 to +12 for EQ) appears on the easily visible main LED display when you turn any knob, providing repeatable settings.

The middle row of controls—marked Amp Model, Warp, and Cabinet—is where the real fun is. The Genesis3 emulates 14 amp types, including sought-after Fender, Mesa/Boogie, and Marshall models, as well as less common amps and generic tube-amp rigs. Settings for vintage fuzz, acoustic-guitar simulation, and Direct (amp-emulation bypass) are included on the Amp Model knob.

One slightly confusing Amp-Model-knob setting, More, accesses three additional preset amplifier types and nine user-programmable HyperModels (which combine the characteristics of two different amp-and-speaker combinations); all other amp models have their own click positions. The User setting selects storage locations for user-designed custom presets. The Cabinet knob offers a similar bounty with 14 flavors of 1×12, 1×15, 2×12, 4×10, and 4×12 emulated speaker configurations. Knob settings for Direct (cabinet bypass) and User (custom cabinet models) are additional options.

The Warp knob, which I haven't seen on any other processor, presents two exciting processes in one. The first process toggles between a green and a red processing channel when you press

down on the knob, enabling instanta-neous A/B comparison of two amp, cabinet, gain, and EQ combinations. When comparing two channels, green or red LEDs encircling the Warp knob clearly indicate channel status, as do single LEDs above the bottom knob row and next to the amp and cabinet selections. A yellow LED at the amp or cabinet control indicates that the same amp or cabinet type has been selected for the green and red channels. When you press the Warp knob, the main LED display also momentarily indicates the selected channel.

When you rotate the Warp knob, a third Warped channel mode is created, in which you can blend the green and red channels in any proportion to create a HyperModel. A numerical readout of the percentage of red channel in the mix (from 1 to 99) appears on the main screen. For guitarists who don't do well with numbers, a more intuitive combi-nation of green and red LEDs around the Warp knob changes as more or less of one channel is dialed in.

The upper portion of the Genesis3 faceplate is a recessed black area in which you access the unit's "brains." All of the controls do double duty, with one function for real-time (Performance
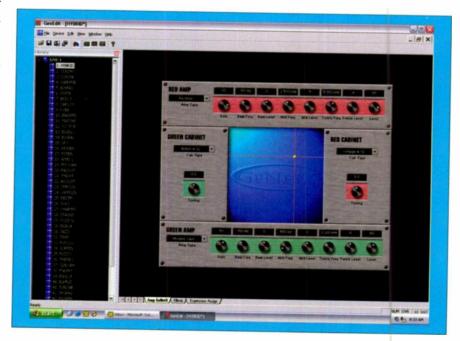
FIG. 3: Several editing features are available only when you use GenEdit, a software editor/librarian for Windows and the Mac OS that's included with the Genesis3.

mode) adjustments and an alternate function for editing operations. Data/Preset, the largest knob in that segment, selects one of 48 presets; the preset name and number are shown prominently on the central LED display. During editing operations, you use the main Data/Preset knob to turn effects on or off and to control other simple toggling functions.

Below the main screen are five knobs: P1 Speed/Amount, P2 Depth, P3 Modulation Level, P4 Delay Level, and P5 Reverb Level. You can use each for real-time manipulation of applicable effects (as on most stompbox pedals) or for changing parameters in the Edit menu. Six red LEDs to the left of the main display show which of the available effects (Pickup/Wah, Compressor/Gate, Whammy/Intelligent Pitch Shifter, Chorus/Modulation, Delay, or Reverb) are active in the current preset.

To enter Edit mode, press the Edit button once and then press it again to scroll through a list of effects. The status lights blink to indicate which effect you're adjusting. As you edit, the main display prompts you by identifying the available parameters and their associated P1 through P5 knobs.

A Tap-It button lets you set the delay time by tapping a tempo in real time. Along with the Edit button, you also use the Tap-It button to name user presets, as well as for utility and expression-pedal-assignment features. Use the Amp Save and Store buttons to save custom presets and HyperModels; press them simultaneously to access the Utility menu.

If you have a computer with a MIDI interface, DigiTech's GenEdit editor/librarian software, which is included, lets you control all parameters and presets in real time as well as perform dumps and backups to a SysEx librarian or MIDI recorder (see Fig. 3). The optional Control X remote foot controller ($229.95) makes it possible to use the Genesis3 as a volume, wah-wah, or Whammy pedal. According to DigiTech, you can also use the pedal to control amp-channel switching, effects bypass, and numerous effects parameters in real time. Because it plugs in to the MIDI In port, however, you can't use it at the same time as external MIDI controllers. I did not receive the Control X for this review.

TEST DATE

There's no need to set the unit on the floor, because it's not a footpedal device. The print around the smaller knobs and indicator lights is too small to read at a distance of three feet or greater. When I set the box on a table-top, I was also frustrated that the height of the P1 through P5 knobs blocked my view of the labels above them. Through trial and error, I found that the best position was angled and at eye level on a sturdy music stand.

Even without cracking open the manual, I was able to scroll through the 48 basic presets and quickly find the appropriate real-time adjustments for each sound. One nifty aspect of the Genesis3 is that it eliminates the need for many single-effect stompboxes by providing chorus, echo, delay looping, harmonization, phase shifting, and a high-fidelity envelope follower in addition to built-in gating. It was definitely a blast to combine effects, amps, and cabinets, and I was immediately impressed by the variety of tones available.

Some straightforward emulations, such as Fender Bassman and Vox top boost, are right on the money. A multitude of thrilling distortion settings also caught my ear, including Carlos, 2Chunk, and Bigduk. A tuned-down preset named Kobb has a grungy vibe to it, although the latency in its pitch-shifted sound became disorienting when I played at faster tempos. I have yet to hear an acoustic-guitar simulation that does anything for me, and unfortunately, DigiTech's take on that is no exception.

As far as the cabinet selections go, I didn't gravitate to any particular favorites, but I could probably find a use for most of the selections eventually. As a fan of lo-fi and twisted sounds, I

Genesis3 GeNetX Specifications

A/D and D/A Resolution	24-bit
Sampling Rate	44.1 kHz
Preset Locations	(48) ROM; (48) RAM
Guitar I/O	(1) hi-Z ¼" TS in; (2) unbalanced ¼" TS out
Digital Output	(1) coaxial S/PDIF
MIDI	In, Out/Thru
Additional I/O	(1) stereo ⅛" TRS audio in; (1) stereo ⅛" TRS headphones out
Frequency Response	25 Hz–20 kHz (+1, −3 dB)
Signal-to-Noise Ratio	>101 dBA
Total Harmonic Distortion	<0.009% @ 1 kHz
Dimensions	10.5" (W) × 2.0" (H) × 6.6" (D)
Weight	2.24 lb.

would like to have heard a single 6- or 8-inch speaker option, as well.

During my getting-acquainted period, I noticed that many presets are more rocked out or effects laden than anything I'd ever use. DigiTech's presets seem tailored more to the tastes of twenty-something shredders than to forty-somethings like me. I yearn for a single effects-balance control or some easy way to reduce the wash of effects programmed into many of the sounds. Combinations of knob twisting and editing can turn individual effects down or off one by one, but there's no way to globally bypass preset effects without the expression-pedal foot controller.

DIVING IN

I knew that more control lurked beneath the surface, so I dove into the 41-page manual to unlock the Genesis3's secrets. The well-written and comprehensive manual patiently guides the reader through the signal processor's basic functions step-by-step before covering the advanced features in depth. A foolproof tutorial for first-time users adds to the document's usefulness. All that's missing from the manual is a clear warning for beginners that because the Genesis3 is a line-level recording device, it can deliver extremely high and potentially damaging output levels when connected to a guitar amp.

Target System Setup mode, under the Utilities menu, is particularly intriguing. It lets you tweak the Genesis3's output characteristics to match the real-world amplifier and speaker cabinet you're using with optional settings for instrument input level, effects-loop input level, or a line-level connection.

Scrolling through the various target systems makes an audible, although subtle, difference in tonal characteristics through an amp, enhancing the unit's effectiveness for recording. While perusing the manual, I discovered many other useful Utilities modes, including a highly sensitive tuner with an A 440 pitch reference that's adjustable from 427 to 453 Hz, stereo or mono output, expression-pedal settings, preset naming, and MIDI operations.

During performance or recording, the benefits of the green/red channel switching are obvious. For example, it was impossible to resist switching between blackface and tweed Fender amps mated with various Fender cabinet combinations. I appreciated the ease with which I could compare subtler aspects of tone and gain settings in real time without having to deal with memory locations or complicated editing. The ability to emulate the tone (though not the noise-canceling properties) of single-coil or humbucking pickups using the Edit menu is another nice touch.

Once I had spent some time devising new sounds, naming and saving them was a snap. You can easily overwrite edited presets to the same number in the separate user bank, which is helpful for

remembering the locations of 48 factory and 48 user presets. Warped amp-and-cabinet HyperModels offer increased options for creativity and saving as part of a new preset.

When I plugged the Genesis3 directly in to a mixer and through studio monitors, I was bothered at first by a consistent buzzing sound as I scrolled through the presets. Then I remembered the Target System Setup, which I had previously adjusted for a 1×12 cabinet. Changing that control to the direct setting made a world of difference in the tones I heard, and the results caught my ear as no direct guitar processor has ever done before.

After another lengthy period of gleeful tweaking, I bowed in the direction of DigiTech's modeling wizards and admitted to myself that it is indeed possible to get great guitar sounds without an amp. Through my studio monitors (Dynaudio BM 15s), I was particularly enamored of the GeNetX Hot Rod

(simulating a Mesa/Boogie Mark II-C), Crunch (an overdriven tube amp), Bassman (a '59 Fender Bassman), and Blues amp settings. Similar devices I've tried have quickly turned me off because of a uniformity of tone or harshness that I couldn't dial out.

I am extremely impressed by the transparency and diversity of convincing amp sounds the Genesis3 has to offer. Turning the bass control up one or two notches often makes the tones even better. The ability to adjust the center frequencies of the bass, mid, and treble knobs is also a big plus. While playing with the Gain and EQ, I noticed that those controls don't always respond immediately to subtle movement, and often a hard twist is needed to "wake up" the Gain and EQ pots.

The Genesis3's output level is sufficient to comfortably drive +4 dBu line-level inputs; noise, though slightly audible, was negligible. Taking the 24-bit digital output and converting it to analog with an Apogee PSX-100 adds noticeable punch and high-end clarity to the tone. The Genesis3 has an adjustable digital-output level in the Utilities menu, which is a big plus in today's studio environment. Creative engineers probably don't need to be reminded that the unit also holds tremendous potential for direct bass and keyboard recording, as well as reprocessing prerecorded tracks. For studio bass tracks, the DigiTech's Bassman emulation is a winner once again, especially with the 1×15 speaker option.

One aspect of DigiTech's guitar processor that does not thrill me is the choice of reverb algorithms. With the exception of the over-the-top coloration provided by the Plate and Church settings, the onboard ambience choices sound generic, metallic, thin, and unconvincing. The spring-reverb emulation is especially disappointing given the status of that sound in the history of electric guitar. I'll just have to hold on to my vintage Fender Pro Reverb for the time being.

SECOND OPINION

To supplement my findings, I enlisted the help of a fellow guitarist with keen ears for the subtleties of tone. John

Finkbeiner, a gifted former student and home recordist, was impressed with the sonic qualities of the Genesis3 through guitar amps and headphones and volunteered that he preferred the sound of the unit to that of the Line 6 Pod and the Tech 21 SansAmp GT2.

To Finkbeiner's discriminating ears, "Cranking the gain and amp level on the Bassman setting yielded the correct progression and type of breakup distortion." That's a hearty thumbs-up coming from the owner of an authentic '50s Fender Bassman amp. He also noted that the unit was a little noisy, but not so much that he was compelled to adjust the onboard noise gate.

Finkbeiner also appreciated the wealth of processing options. He was particularly intrigued by the cabinet-tuning feature, which lets you adjust the resonance of selected cabinets one octave up or down. Like me, though, he thought many of the reverb and echo presets were excessive and was quickly annoyed that he couldn't bypass the Genesis3 circuitry without the Control X expression pedal.
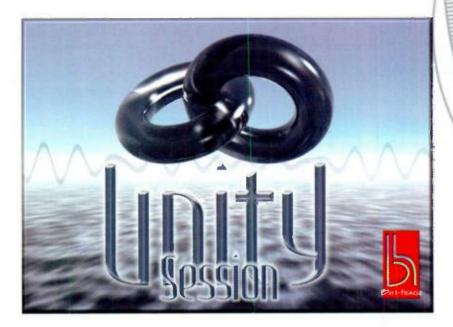
REVELATIONS

Though it offers a complex roster of features, the DigiTech Genesis3 is remarkably easy to use, thanks to an instructive display, intuitive controls, and a clear manual. As a busy studio professional, I certainly value the rich variety of credible guitar sounds that you can access immediately, not to mention that the unit's deeper editing capabilities can be mastered in an afternoon.

As a guitarist and bassist with a few decades of experience under my belt, I have to hand it to DigiTech for making a box that's simply a heck of a lot of fun to play around with. The Genesis3 definitely raises the bar for quality and value in modeling devices. At just over two pounds, it sure beats hauling a truckload of old amps to the gig!

Guitarist **Myles Boisen** *(mylesaudio@aol .com) has worked with David Lynch, Tom Waits, John Zorn, Fred Frith, Nina Hagen, and many others. He is also head engineer at Guerrilla Recording and the Headless Buddha Mastering Lab in Oakland, California.*

Obsession.

A B L E T O N

LIVE 1.1 (MAC/WIN)

The real-deal, live-loop mixing machine.

By Erik Hawkins

For some time I've dreamed of owning an application that would combine loops in real time without stopping playback to reload, like an automatic pistol that could fire loops in a never-ending stream of remixed beats and sounds. After trying all of the DJ remix programs I could find and attempting to coax my digital audio sequencer into acting like a loop-remixing program, I was still hauling my hardware sampler to gigs.

Every software-based setup fell short in crucial areas. Among the functions I couldn't find were flexible loop triggering, multiple outputs, sample auditioning on the fly, and MIDI. In addition, RAM-based playback limited the number of samples that could load simultaneously. The only setup that even came close to giving me the kind of dynamic remixing flexibility I wanted was a hardware sampler with multiple outputs and a 6-channel DJ mixer. However, that system still paled in comparison with what I believed should be possible with a laptop and the right software program. (Besides, carrying around all that equipment was killing my back.)

Then a DJ friend told me about Live, the new program for mixing loops and samples in real time from Ableton, a German software company. In January I made it a point to catch a demo of Live at the NAMM conference. I expected to be disappointed but ended up hardly believing what I saw.

Ableton Live has all of the features I've been dreaming of. You can audition loops and throw them into the mix on the fly, and effects plug-ins can be treated in much the same way. A prelistening feature lets you cue samples before sending them to the main mix, and Live can address multioutput sound cards. Live plays back audio streamed from your hard drive, so running out of RAM isn't a problem. Live even has MIDI to let you use an external controller. As soon as I could, I obtained a copy of Live and gave it a thorough examination.

Minimum System Requirements

Live

MAC: G3/233; 128 MB RAM; OS 8.6; CD-ROM drive

PC: Pentium/300; 64 MB RAM (128 MB recommended); Windows 95/98/2000; CD-ROM drive; Windows-compatible sound card

EASY INSTALLATION

Live runs on Macs and PCs, and both versions ship on the same CD-ROM. (Mac users will be interested to know that Live is the first multitrack recording application that runs on OS X.) I used Live on a Power Mac G4/450 MHz with 704 MB of system RAM, lots of drive space, and a Digidesign Pro Tools/24 Mix card. In addition to the minimum system requirements, you should have plenty of hard-drive space to store sample libraries (you'll want them readily available while you're mixing) and an ASIO-compatible sound card for the best possible fidelity. The software works with Sound Manager on the Mac and with DirectX-compatible sound cards on the PC, but those output sources might not provide enough bottom end to really make your mixes bump.

Installing Live is simple, and with no extensions to install, there's no danger of driver conflicts. Authorization is through challenge and response. Live generates the challenge after you enter its serial number. Then you can register the product on Ableton's Web site to obtain an immediate response. If you can't get online right away, the program runs for ten days before requiring the response.

LIVE IN PERSON

Everything in Live is accomplished in a single window that is divided into three sections. The largest area toggles between Session View (see **Fig. 1**), an audio clip mixer where you will spend most of your time; and the Arranger (see **Fig. 2**), a sequence editor that features grids and blocks to display audio events and control data. To the left of the main area is the Browser, and below that are the Detail Area, which toggles between

FIG. 1: Just beneath the Session View, the Clip View for the currently selected mixer channel is shown. A list of the included plug-ins appears on the left.

FIG. 2: You can edit your recorded performance in the Arranger View. Below that, the Track View for the currently selected mixer channel shows controls for some of Live's plug-in effects.

three displays: the Clip View, which shows a sample's attributes; the Track View, which shows a channel's inserted effects chain; and the Bus View, which allows you to monitor the mixer's buses.

You can also display the program's handy help window, called the Info View, below the Browser. When you hold the cursor over an element, its description appears in the window; that invaluable feature almost replaces reading the manual.

Around the sides of the main window are various screen-access controls and indicators. At the top of the window are indicators for CPU load (calibrated by percentage), MIDI In, and hard-disk overload. All are extremely useful, but the hard-disk overload indicator would be more valuable if it were a drive-usage meter; it doesn't turn on until it's already too late and audio playback has probably been disrupted. You can horizontally resize the Browser so that you can read long file names. To optimize screen space for the Session View/Arranger, you can hide the Browser, the Info View, and the Detail Area (see Fig. 3).

Live's default appearance is mostly gray with lime green highlights. Thankfully, Live includes a few alternate skins that feature different color schemes to spice up the program's appearance.

BROWSING FOR A BRUISING
The Browser is logically organized, with buttons in its left margin for opening the sample and effects folders on your

hard drive. Four of those buttons—one button for VST plug-ins and three for sample libraries—are user-defined and require you to specify the locations of the folders they open. Another button is dedicated to the program's built-in effects. The buttons can save a lot of time because they let you jump directly into several different folders without the need to continually navigate your drive's folder hierarchy.

Live recognizes WAV and AIFF audio files; if a folder doesn't contain those file types, the folder appears empty in the file dialog. I wish that Live recognized some additional file types, especially Sound Designer II (SDII).

In the Browser's prelistening bus, you can audition samples at your session's current tempo without having to drag them into Live's mixer. Although the sound quality isn't as good as playing the sample in Session View, the ability to cue samples directly from your hard drive before dragging them into the mixer is marvelous.

PLAYING THE SLOTS
In Session View, you can play back samples from channel slots. After you drag a sample to a slot, the length of the sample governs the amount of time it takes to load; an 8-bar loop at 130 bpm takes about a second.

A channel can contain as many slots as you want, and the number of mixer channels is practically unlimited. However, the number of samples you can play back simultaneously depends on your computer. My computer could play no more than 12 simultaneous loops with 8 plug-ins that included a long reverb and a medium delay. To understand Live's mixer channels and slots, just think of each channel as a monophonic sampler.

A display beneath each channel's slots shows the length of the sample that's playing, how many times it has looped, and a pie chart that indicates its progress; as the sample plays from

beginning to end, the pie fills up. That information is quite useful, but the display is painfully small; some means to enlarge it would help.

MASTERFUL MIXER
Each mixer channel features a fader, a pan knob, two send knobs, and an input-source selection menu. You can hide any of those elements, leaving only the sample slots visible. You can have as many as four sends, and each send return supports drag-and-drop effect insertion, just like the regular mixer channels. All that's missing on the mixer is a solo feature; although you can use the prelistening bus as a solo when it's assigned to the same output as the master, it's not quite the same as having a discrete solo button on each mixer channel.

Input sources, in addition to channel slots, can be routed from Live's main output, your sound card's input, or Propellerhead Reason and ReBirth programs. When you route Live's main out to a channel's input, you can sample your own mix and then drag that sample back into the mix without stopping playback (presenting some interesting possibilities that I will discuss in a minute). With your sound card assigned as the input, you can use Live just like a multitrack hard-disk recorder but with greater flexibility in terms of sample tweaking and looping in real time.

If you open Propellerhead's Reason or ReBirth in the background, you can route their audio to Live's mixer using the ReWire system extension. ReWire also synchronizes the programs, with Live serving as the master clock. I ran all three programs at the same time and was pleased with the results. Starting playback on Live caused Reason and ReBirth to begin playing also. The ability to use those two programs in tandem with Live opens up a new world of soft synthesis, virtual drum machines, and step sequencers that you can integrate seamlessly with Live. Ableton really did its homework.

TAKING THE BUS
Live's mixer is equipped with a wonderfully flexible bus system. You can

Breaks sound (and price) barriers.

route each channel's audio to the master bus, the prelistening bus, or a sound card's multiple outputs (in stereo pairs). Each channel has dedicated buttons for sending its signal to the master or prelistening buses. In the program's Preferences page, you have the ability to set the master and prelistening buses to any of your sound card's outputs. Assuming that your sound card has multiple outputs, you are able to select those outputs in each channel's pop-up output-assignment menu. You can do all your busing during playback with no interruption in audio—just like a real mixing board.

Live's busing architecture blows other remixing applications away. Prelistening lets you use Live with a four-output sound card (two stereo cards might work as well) to perform DJ-style cuing. Connect the outputs assigned to the prelistening bus to your headphones and then plug the outputs assigned to the master bus to your house speakers. Use the prelistening bus to privately cue loops over your headphones before sending them to the house speakers. That's the same method DJs use with two turntables and a DJ mixer— except they must also beat-match their tracks, and Live does that automatically.

Another alternative is to simply bus each mixer channel to its own stereo output (which should require a sound card with at least eight outputs). Plug the outputs in to a DJ mixer with lots of inputs and use the board's cue system instead. That setup provides more immediate control over levels because you don't need to fuss with Live's virtual faders. Use the board's faders to control individual output levels. Such a setup also gives you easy access to outboard gear, such as external effects units or even the DJ mixer's own EQ, to lighten your computer's processing load. That's how I have Live set up, and I'm extremely pleased with the amount of flexibility it brings to my real-time remixing endeavors.

EFFECTS GALORE

It is possible to use VST effects with Live, and the program also comes with its own set of effects plug-ins. Stock ef-

fects include standbys such as chorus, compression, EQ, and delay, as well as more uncommon effects such as vinyl distortion and Erosion. An effects chain is created by dragging and dropping plug-ins into a channel's Track View. You can have as many effects chained together as your computer can handle. It is possible to drag effects into Track View during playback, but any plug-in that requires a great deal of digital-signal-processing power (such as long reverbs and delays) might cause dropouts. It is probably a good idea to have all of the effects in place ahead of time so that you do not risk overloading your CPU by dragging in new effects during a mix.

The user interface for most of the included effects and for all VST effects allows you to control two parameters at the same time by moving a cursor inside a field where the x- and y-axes are each tied to a different parameter; manipulating parameters in that fashion is ideal for live performance. The parameters are fixed in Live's plug-ins, but for VST plug-ins they are user-definable. The trade-off is that VST plug-ins look different in Live than they do in other programs. You can recall a VST plug-in's original interface, but in its own window outside of the Track View. You can automate practically every effects parameter and assign a MIDI continuous controller to most controls, which is nice.

Live's stock plug-ins are decent and seem particularly suited for composing electronic and loop-flavored music— the type of music you're most likely to compose with Live. Among my favorite plug-ins are the Auto Filter, which is great for creating hardcore filter sweep sounds, and Filter Delay, a tempo-based delay that is useful for creating interesting rhythmic backdrops out of mundane-sounding loops. There are several additional delays that can be locked to tempo, but beware: they consume a lot of processing power.

FIG. 3: Shown with all the other windows hidden, the Session View turns orange and purple when you apply Live's Disco skin.

LOOP AND TRIGGER

It's not enough for a real-time remixing program to simply play back beat-matched loops; to make it work in live performance, it needs serious sample-triggering and loop-tweaking options. Live doesn't disappoint in either area. Every sample loaded into Live can have its own triggering setup, tuning, gain, loop points, and Warp Markers (which determine how Live's automatic time-stretching algorithm is applied). You can even save a sample's settings so those parameters are instantly recalled as soon as you load the sample into a channel's slot.

You can trigger samples with your mouse or QWERTY keyboard, or by playing a MIDI note. There are four ways to trigger a sample: a Note On message begins playback; a Note On message begins playback and a Note Off message stops it; a Note On message begins playback and another Note On message stops it; and holding a note repeats the first portion of the sample (the length of which is determined by the sample's first Warp Marker and the session's quantize setting). The last mode is cool because after setting your session's quantization value to 16th notes, you can repeat the first beat of a sample to play 16th-note fills.

I'd like to see a trigger mode in which you can start a sample's playback from any Warp Marker without losing the loop's lock on the downbeats; Mixman's Mixman Pro software offers that type of triggering. It would also be much cooler if Live's samples responded to Velocity.

A loop can be continuous or one-shot, and you can set its length to any number of bars. The sample's start point can be anywhere within the sample as long as it isn't after the beginning of the loop. A waveform overview (in the Clip View) lets you drag the sample's start point and loop-in and -out markers into position; if you prefer, you can enter marker locations directly into the appropriate text fields.

GO TO WARP

Warp Markers are similar to the beat markers of a REX file, but instead of providing directions on how to cut up a sample to match a project's tempo, they tell Live how the sample should be time-stretched to match the project's tempo. Divisions for Warp Markers range from whole bars to 32nd notes. Live automatically applies Warp Markers to a sample according to its length and specified time signature.

If the sample is trimmed to a perfect loop and its rhythm doesn't waver too much, automatic Warp Markers usually work fine. However, when a sample's tempo is erratic, you can freely move Warp Markers around in the waveform overview to manually align them to the waveform's beats. Live's time-stretching algorithm is then automatically applied to the sample during playback.

Live's time-stretching methodology is similar to that of other loop-based programs such as Sonic Foundry Acid and BitHeadz Phrazer. If you're used to working with those programs, Live will probably be familiar territory. I'm more accustomed to working with beat markers in the style of Propellerhead ReCycle, so Live's system threw me for a loop at first. After learning the Warp Markers, though, I found them pleasingly accurate, and the time stretching sounded good. The program won't perform miracles, of course, but most material with tempo changes of 25 beats or less sounds okay in the mix.

ROUGH ARRANGEMENT

After recording a performance using Live's onboard sequencer, you can edit sample-trigger locations and automation data in the Arranger. At first glance, the Arranger appears to be missing most of the features found on full-fledged MIDI sequencers such as Mark of the Unicorn Digital Performer and Emagic Logic Audio. Although the sequencer is not comprehensive, it does offer the essential commands necessary for basic editing. Live's design is so unlike those of more popular sequencers that it's somewhat counterintuitive.

Initially tracks are displayed in a mode that allows only limited editing. Blocks of performance data can be dragged vertically within a track, dragged horizontally between tracks, and deleted. To edit a track's performance in detail, you need to unfold the track. Unfolding a track displays its waveform and lets you cut and split blocks of performance data. You can also adjust a block's start and end points by expanding or contracting its edges. When a track is unfolded, Live can display its level and pan moves (or any other controller you have selected) as break-point automation data.

Editing commands are located in the main menu bar's Edit menu. There's no editing-tool palette, but adding one might make the Arranger much more intuitive. Some kind of draw tool would make it easier to add events to a track.

Navigation and zooming are accomplished using a small box that overlays the sequence overview bar. Drag the box to navigate the sequence; drag its edges to zoom in and out of the Arrange display. Getting around a sequence and zooming with that tool are particularly tedious—an old-fashioned scrollbar and magnifying glass tool would be better. (For another perspective, see Todd Souvignier's loop-sequencer survey, "Loop-a-palooza," on p. 50.)

GET YOUR MIX ON

You can quantize your performance on input so that when you trigger a sample, the event snaps to the nearest note value, which is determined by the quantize setting. Quantization can be set from whole bars to 32nd notes. For my needs, 16th-note quantization provided the best results, because I could trigger samples at almost any point, not just on the downbeats. You can also turn off quantization, which is good if you want more precise control over when a sample is triggered and, ultimately, how it sits in the groove.

Triggering samples with the computer's mouse or keyboard does not excite me; using a performance-oriented MIDI controller is much more inspiring. Every channel's slots can be assigned a MIDI note number. An entire horizontal row of slots, called a Scene, can also be given a note number. When you arrange a session's samples to trigger on different notes of your controller, using Live becomes more like playing an instrument than simply tinkering with a computer program. With careful planning, you can even play chords to launch groups of prearranged loops.

Once several cool-sounding loops are playing together, Live can bounce down the arrangement into a stereo WAV file. Without interrupting playback, Live will then automatically loop the new sample and load it into a slot on the same channel that was used for recording. That amazing feature opens up myriad possibilities. For example, you never need to worry about running out of processing power; just bounce down when you

near maximum CPU usage, clear out your slots, and start fresh. The potential for an endless mix is at hand.

INTO THE NIGHT

I had no real problems using Live 1.1; it even played through clusters of CPU overload peaks without crashing. Live gave me little to complain about, which is surprising considering that it was introduced so recently. Sure, the mixer is missing a solo feature, sample triggering isn't Velocity sensitive, the included plug-ins could use a few more effects, and the sequencer is rough around the edges. Still, those are minor shortcomings when you consider all the features that are packed into Live.

At press time, Ableton is on the eve of releasing Live 1.5. The updated version will run as a ReWire slave in tandem with popular sequencing programs as well as ReBirth and Reason. In addition, it will sync to external MIDI Clock and provide a master clock source for syncing other programs and external devices. A new Render-to-Disk function

will quickly export Live's audio output to other programs. Other enhancements will include a new reverb plug-in and less strain on CPU resources.

Between its various trigger modes, drag-and-drop operations, busing and prelistening features, ASIO support, MIDI control, VST effects plug-in compatibility, and Reason and ReBirth integration, Live has the bases covered. Incidentally, Live also supports MIDI Clock and MIDI Time Code, so you can

lock your computer up with your friends' systems for beat-perfect remixing jams. At the price, that's darn impressive. Version 1.1 is such a robust foundation to work from, I can't wait to see how Live evolves.

Visit **Erik Hawkins**'s *fledgling record label (www.muzicali.com) to hear music made with the latest studio gizmos and to buy his virtual-recording-studio book,* Studio-in-a-Box *(www.artistpro.com).*

PRODUCT SUMMARY

Ableton
Live 1.1 (Mac/Win)
sample sequencer
$299

FEATURES	4.0
EASE OF USE	4.0
QUALITY OF SOUNDS	4.0
VALUE	4.5

RATING PRODUCTS FROM 1 TO 5

PROS: Automatic beat matching. Good time-stretching algorithm. Resampling. Almost total automation. ASIO and ReWire support. Assignable outputs. Flexible bus architecture. MIDI control. Drag-and-drop operation. VST compatibility.

CONS: No solo buttons. No Velocity sensitivity. CPU-intensive delay effects. Limited sequence editor features. Awkward Arranger navigation.

Manufacturer
Ableton/Midiman (distributor)
tel. (800) 969-6434
e-mail info@midiman.net
Web www.midiman.com

C L A V I A

NORD ELECTRO

Electromechanical keyboards of old are back without the backache.

By Nick Peck

nce upon a time, musicians sporting sideburns, large zodiac medallions on thick gold chains, and tight polyester jumpsuits grunted and heaved their monstrously heavy keyboards onto the world's stages. They begged and cajoled their temperamental behemoths to survive another gig without breaking tines, reeds, or Leslie motors. The mighty Hammond organ, Rhodes and Wurlitzer electric pianos, and Hohner Clavinet were large, heavy, bulky, and often in need of maintenance. But the sound that emanated from them was glorious.

Clavia, manufacturer of the Nord series of virtual analog synthesizers, has responded to the thirst for vintage keyboard sounds by creating the Nord Electro. The Electro is a virtual electromechanical instrument that simulates a Hammond B-3 organ, a Hohner Clavinet, Rhodes and Wurlitzer electric pianos, and a Yamaha CP80 portable piano. There are no synthesizer sounds, and programming capabilities are limited to tweaking the effects and Hammond settings. The Electro is incredibly compact and lightweight, has a great keyboard action, and delivers credible simulations of several instruments.

FORM FACTOR

The Electro was designed to be highly portable (see **Fig. 1**). The 61- and 73-key versions are not much taller than the height of the key bed. (A rackmount version, the Electro Rack [$1,699], is slated to be released by the time you read this.) The case is the standard Nord fire-engine red, and the buttons, knobs, and distinct lack of an LCD are the same as on others in the Nord series. In a nod to its '70s progenitors, Clavia clad the sides of the Electro in red-stained wooden end blocks—a nice touch. The Electro has an internal power transformer as well as a two-prong power cord rather than the ubiquitous three-prong grounded IEC power cord that is found on most pro-audio and synth components.

The controls and displays are tightly grouped above the keys. In addition to ¼-inch stereo outs, a headphone jack, and MIDI In and Out, the rear panel sports jacks for an assignable control pedal, a Sustain switch, and a switch to change the simulated Leslie's rotor speed (see **Fig. 2**). When new instrument samples become available, the Electro's USB port will facilitate downloading them from Clavia's Web site. A second pair of outputs would have been most welcome, allowing the organ section to run into a Leslie and the piano section to run into an instrument amp.

I've never been wild about Nord's bright red look. It is distinctive and edgy, but it can distract from the performer during a gig. I wish Clavia would offer a black model with silver writing as an alternative; it would fit perfectly into a rig that includes vintage keyboards.

ELECTRO INTERFACE

The front panel is a model of efficiently organized elegance, with dedicated knobs and buttons for every commonly used feature. On the left is the area for program storage and selection as well as system-setup functions. A two-digit LED display shows the preset and system values, but users would be better served with a small LCD. The octave-shift buttons, which let you transpose the pitch up or down three octaves, are especially useful on the 61-key Electro. To their right are the instrument controls.

The Electro is conceptually grouped into two instrument sections: organ and piano. The organ is a physical model, and the pianos are Velocity-switched multisamples. You choose between the organ and piano sections using an instrument-select button.

The organ section features the most frequently used tone-shaping controls found on a Hammond B-3. Buttons are provided for percussion (On/Off, Second/Third Harmonic, and Standard, Soft, or Slow Decay), vibrato (simulations for the B-3's three levels of vibrato and three levels of chorus), and nine sets of drawbars.

The Electro's drawbar buttons substitute for the drawbars on a real Hammond organ; their tone-shaping ability is a critical aspect of B-3 performance practice. Each virtual drawbar represents one partial of the organ tone, letting the player change the sound markedly by moving drawbars on the fly. Each drawbar consists of an up and a down button and an eight-segment LED display for each partial.

The advantages of the Electro's drawbars are that partial settings are instantly updated during program changes, and real drawbars stand a greater chance of eventually breaking. On the other hand,

FIG. 1: At 20 pounds or less, the Nord Electro is a playable, lightweight alternative to dragging around a Hammond B-3, a Leslie cabinet, and a collection of electric pianos.

the drawbar buttons aren't as direct and immediate as real drawbars, and they will take some getting used to on the part of the seasoned Hammond player.

On the organ instrument, you can split the keyboard to simulate a Hammond's upper and lower manuals. The split point is fixed and allows enough room in the lower manual for left-hand chord comping or walking bass. As an alternative, you can plug another MIDI keyboard into the Electro to operate as the lower manual.

Although I applaud Clavia for assigning a separate control knob or button to each primary function, it missed the boat in one area: the positioning and size of the Leslie-simulation buttons. On a real Hammond, the Leslie Speed switch is on the organ's far-left side to permit a quick swipe of the left hand to change rotor speed. The Electro places the switch toward the right side of the front panel, and the button is no larger or easier to reach than the rest of the buttons

FIG. 2: The Electro's rear panel offers two unbalanced audio outputs, a stereo headphone jack, and two MIDI ports. Three jacks are provided for connecting footpedals and switches, and a USB port makes it possible to download new sounds and operating-system upgrades.

on the instrument. However, you can control the speed with a footpedal by means of the rotary speed jack.

The piano section is straightforward and features a button that switches between the Rhodes Mark II Stage Piano, Wurlitzer 200A, Clavinet D6, Yamaha CP80 Electric Grand, acoustic grand (Ac.Grand), and Option (for future use) multisample sets. Additional presence controls in the piano section let you dial in a midrange bump or cut to shape the multisample's tone a bit.

ACTION AND PLAYABILITY

I am a snob when it comes to synth keyboard actions. I wondered how any man-

ufacturer would handle an action that covers the needs of organ and electric-piano performance. Clavia has succeeded marvelously. The keyboard uses waterfall keys similar to those on real Hammonds; they have no lip, and the front edge reaches down into the key bed, unlike the keys on most synths. The action has a bit of bounce, weight, and stiffness, which is great for lightning-fast repetition of single notes, a technique common to Hammond playing.

The action is heavier than but certainly in the ballpark of a Clavinet's action. Although it's definitely lighter than the thuddy, sluggish feel of a Rhodes or the pianoesque action of a CP80 or

Wurlie, the Electro's action is perfectly acceptable for idiomatically playing electric-piano sounds as well as organ.

ELECTRO SOUND

For years I have owned and played all the instruments simulated by the Electro. I'm comparing the Electro directly with them, not with instruments that simulate the real thing. In that context, the Electro's sounds are a mixed bag, ranging from excellent to, well, not; in all fairness, though, I feel the same way about many other vintage keyboard simulations.
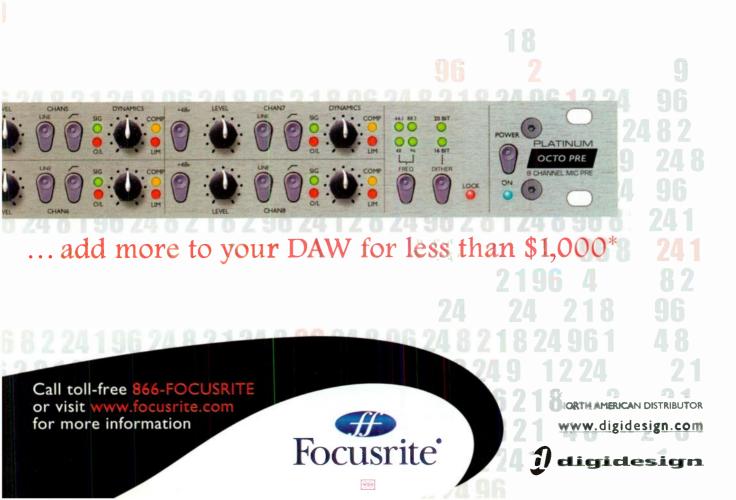
Hammond. The Hammond model sounds wonderful. The tone partials are thick and fat, with a tremendous, tight low end and shimmering highs. Dialed-in drawbar settings sounded exactly as I expected them to; they behaved similarly to my real Hammonds. The percussion is good and usable for bringing out ripping lines over a band. The chorus and vibrato are okay—not dead-on,

but good enough. The Leslie simulation is one of the best I've heard; it is musical and crystalline, and the changes in motor speed sound amazing.

The one glaring problem in the Electro's Hammond is the key-click simulation. It is a sharp, spiky, bright click at the beginning of each note that is like nothing I have heard in any Hammond. It sounds unmusical to my ears, and you can't defeat it or lower its volume. Fortunately, there is a work-around: keep the Leslie simulation on while using the Hammond model, and the lowpass filter in the simulation completely removes the key click. Clavia reports that a software patch that allows the user to defeat the key click should be available soon.

Rhodes. The Rhodes multisample is based on a Mark II Stage Piano. The samples and Velocity crossfading are great. The instrument is full and rich, though just a shade bright and bell-like for my taste. With a nice dynamic range, the

sampled Rhodes really honks and bites when you dig in. Throw on a touch of the phaser or chorus effect and a little overdrive, and you're in business. This instrument is totally musical.

Wurlitzer. At first I didn't like the Wurlie, but it's growing on me. The tone is rich and smooth, but it just doesn't have quite the growly honk of my beloved 140B. Something is missing in the subtle dynamics of the instrument; perhaps it's a bit too clean and polite. The tremolo effect helps a lot, though, and the instrument sounds good.

Clavinet. The Clav is a disappointment. The sample Clavia chose is bright and fairly flat, without any of the ringy honk and beef that characterize the real thing. Clavia didn't capture the subtlety of the sound, the clangorous overtones, or the grittiness that make songs such as Stevie Wonder's "Superstition" and Billy Preston's "Outa-Space" classics. Running the Clav sound through a real wah-wah pedal helps, but a more nasal, midrangey

sample set would be a great thing to add in an upgrade.

CP80. The CP70 and CP80 were Yamaha's "portable" electroacoustic pianos. Popular in the 1980s, those pianos used a small soundboard and electric pickups to create a metallic, biting sound with little low end. The Electro achieves a credible simulation. Throwing a little chorus on the instrument creates a pretty-sounding alternative to the Rhodes, making it useful for ballads.

Acoustic piano. As a bonus, Clavia includes one nonelectromechanical simulation: an acoustic grand piano. Acoustic pianos are difficult instruments to capture, and Clavia did not succeed in the Electro. The piano is thin and two-dimensional. Clavia would have done better to use the space for an al-ternate Clavinet sample set. I hope that the acoustic-piano set can be upgraded in the future through the Electro's USB port. (According to Clavia, that will be possible by the time you read this.)

EFFECTS SECTION

The effects section is a serial chain of six effects stages, each with its own On/Off button and control knobs, making it simple to quickly dial in the right sound for your application. Effects include the aforementioned midrange presence control for the piano section, a Modulations stage with six amplitude- and filter-based effects, an Effects stage with six pitch-based effects, an Overdrive stage, the Rotary Speaker stage, and a final high- and low-shelf EQ stage.

The Modulations stage includes a

Nord Electro Specifications

ORGAN SECTION	
Sound Engine	physical modeling
Polyphony	61-note or 73-note with internal keyboard; 146-note with additional external MIDI keyboard
Virtual Drawbars	(9) drawbars; (9) user drawbar settings; (9) preset drawbar settings
Keyboard Splits	lower manual/upper manual; preset split point
Effects	swell; percussion (2nd, 3rd, loud, soft, slow, fast); chorus/vibrato; overdrive; rotary speaker simulation; organ morph
PIANO SECTION	
Sound Engine	sample playback
Polyphony	24-note
Simulations	Clavinet D6, Wurlitzer 200A, Rhodes Mark II Stage Piano, Yamaha CP80 Electric Grand, Ac.Grand, Option
Effects	presence
GENERAL	
Keyboard	61-note or 73-note; Velocity-sensitive
Display	2-character LED
Global Effects	overdrive, EQ, chorus, flanger, phaser, tremolo, wah-wah, autowah, panning, ring modulation
Audio Outputs	(2) unbalanced ¼" TS; (1) ¼" stereo headphone
Control Inputs	(1) assignable footpedal; (1) Sustain switch; (1) Rotary-Speed switch
MIDI Ports	In, Out
Memory	48 RAM programs
Dimensions	61-note: 35.8" (W) × 3.7" (H) × 11.6" (D); 73-note: 44.0" (W) × 3.7" (H) × 11.6" (D)
Weight	61-note: 17 lb.; 73-note: 20 lb.

tremolo that is essential for dialing in a convincing Wurlie sound, an autopan good for a Rhodes patch running in stereo, a ring modulator, and a decent wah managed by a control-voltage footpedal for the Clav. The Effects stage consists of two flangers, two phasers, and two choruses, all of which are appealing but not as good as high-end analog effects pedals. Similarly, the Overdrive stage is an okay distortion simulation, but a good overdrive pedal would sound better. The Rotary Speaker, however, is terrific. You can use it on the piano and organ sections, which is a great touch. Like other units in the Nord series, the Electro offers no reverb or delay effects.

ELECTRIFIED

Clavia has created an interesting product in the Nord Electro. The Hammond simulation is solid and credible, though the drawbar buttons take some getting used to. The sampled pianos are a mixed bag, but the size, weight, action, and built-in effects make the instrument a godsend to the gigging retro keyboardist.

Does the Electro sound exactly like the instruments it tries to simulate? Nope. Does it sound musical? Yes, for the most part. For gigging musicians who focus on vintage keyboard sounds, the practical convenience can't be beat. I have yet to find an instrument that can take the place of my beloved vintage instruments, but in a gig or rehearsal situation to which lugging the real thing is inconvenient or impossible, I would throw the Electro on top of a real Wurlie and know my bases are covered handily.

Nick Peck creates sound for film and games (www.perceptivesound.com) and plays keyboards in the Bay Area jam band Ten Ton Chicken (www.tentonchicken.com). His e-mail is nick@tyedye.com.

PRODUCT SUMMARY

Clavia
Nord Electro
electronic keyboard
61-key model
$2,099
73-key model
$2,299

FEATURES	4.5
EASE OF USE	5.0
AUDIO QUALITY	4.0
VALUE	3.0

RATING PRODUCTS FROM 1 TO 5

PROS: Excellent Hammond, Rhodes, and Wurlitzer sounds. Compact and lightweight. Terrific keyboard action. Clear and comprehensive user interface. Convenient electromechanical keyboard simulator for gigging keyboardist.

CONS: Expensive. Thin Clavinet and piano sounds. Hammond key click is too loud and bright. Organ performance-button layout is too different from real Hammond.

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B L U E

BABY BOTTLE

If you're looking for smooth, full mids, this Baby's for you.

By Sean Carberry

I n 1992 the sorcerers at Blue secretly designed a line of large-diaphragm condenser microphones so distinctive that they would shock and amaze the audio community. During the past decade, the mad scientists slowly and meticulously crafted these mics from the ground up, giving each a distinct sonic personality as well as an unusual appellation such as Blueberry, Mouse, Kiwi, or Dragonfly.

FIG. 1: The Baby Bottle uses a minimalist circuit design and hand-built capsule to capture a rich, natural midrange without hyped highs and flabby lows. The mic is shown with the optional shockmount and pop filter.

One by one, the company unleashed the microphones on a public unaccustomed to the unorthodox. Initially apprehensive, the audio world soon welcomed these friendly creatures into its midst. In fact, the members of the community warmed to the misfits to such an extent that they have since bestowed award after award upon Blue microphones.

Now that the public has accepted and come to love these sometimes bizarre-looking but always great-sounding microphones, Blue has released the final product in its esteemed line of transducers. The Baby Bottle completes the series of Blue large-condenser mics, and as the least expensive of the Blue line, it provides an apt bookend to the pricey patriarch of the family, the revered Bottle.

FATHER'S FOOTSTEPS

Like the Bottle, the Baby pays cosmetic (and sonic) homage to Georg Neumann. While Dad resembles the stout Neumann CMV 3, Junior looks like the svelte CMV 563. The Baby Bottle's lollipop capsule stands authoritatively atop its slender, cylindrical body. However don't try to remove the head—though the papa Bottle provides movable and interchangeable capsules, the Baby Bottle's lollipop capsule comes hardmounted.

Unlike the lollipops on some of Blue's other mics, the Baby Bottle's lollipop is spherical and about the size of a Ping-Pong ball. The industrial-strength grille houses the handmade, single-membrane, fixed-cardioid capsule. A textured, metallic black coating covers the microphone body, which is otherwise adorned with a brass Blue emblem on the front and the Baby Bottle nameplate on the back. The mic comes with a swivel-mount threaded to the base. Optional accessories include the Baby Shock shockmount and Baby Pop pop filter (see **Fig. 1**), which are sold together for $149.

Like all Blue mics, the Baby Bottle ships well protected; it comes in a beautifully crafted coffin-type cherry-wood box with a tongue-and-groove sliding lid. I unraveled the bubble wrap and discovered a faux-velvet bag enshrouding the mic. I was initially concerned about the lack of padding in the box, but the folks at Blue explained that I had an early model of the box and thus did not receive the die-cut foam bed that cradles the mic.

INNER BEAUTY

Don't let the price fool you—the Baby Bottle is no corner-cutting budget microphone. According to Blue, the sound the company imagined for the Baby simply lends itself to a less expensive design. The goal was to create a mic with an unhyped, unaffected sound. So Blue did away with the extensive equalization found in most modern condensers, creating instead a capsule with a neutral character. One inch in diameter, the capsule comprises a 6 µm thick gold-and-aluminum-sputtered membrane mated to a simple Class A, transformerless output circuit. The result is a clean sound with the lowest self noise—5.5 dB—ever (not) heard from a large-diaphragm condenser, let alone a budget model.

Like all Blue mics, the Baby has no pad or filter. That is in keeping with Blue's philosophy of simple, clean electronics. It also has the highest output level I've ever experienced from a microphone. That can be a plus in a home-recording environment, especially if you have noisy mic preamps; in professional recording studios, however, I had to use pads more often than I would have liked.

RIGHT IN THE MIDDLE

The sound of the Baby Bottle not only belies its price tag, it also breaks with the budget-microphone paradigm. People have been beaten to deaf by inexpensive Neumann U 87 clones, most of which are aggressively, often painfully, colored. I hate the trend of abrasive-sounding mics (not to mention the "pointy" sound of many contemporary CDs).

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KORG

With the Baby Bottle, Blue avoided that sound altogether. Rather than emphasize top and bottom, the Baby Bottle permits the midrange to speak. This microphone simply sounds smooth and full in the mids without any drastic peaks or valleys. Because Blue didn't load the output electronics with EQ circuitry, you don't hear any of the inherent phasiness of hyped or highly colored microphones. Again, the idea behind the Baby was to let the source speak for itself rather than allow the mic to do the talking. I dig that approach. (It's also the reason I love ribbon mics so much.)

BROADCAST NEWS

Upon receiving a pair of Baby Bottles for review, I plugged them in at WBUR, the National Public Radio station in Boston for which I work as technical director. Normally, all on-air mics at the station are Neumann U 87s. The U 87s work great on some voices, but

not as well on others. Compared with the 87s, the Baby Bottles sounded more natural and less peaky—and a boatload louder. The Baby Bottles were full, present, and smooth sounding; in comparison, the U 87s sounded somewhat artificial on some voices.

The Baby Bottle also exhibited a slightly wider pickup pattern, which afforded a little more leeway in placement and allowed the talent to move around more without dire consequences. Although the Baby exhibited excellent off-axis rejection at 90 to 135 degrees, at 180 degrees the mic picked up more information than most large-diaphragm cardioid condensers I have used. In addition, because the Baby has less bottom end than other large-diaphragm condensers, you can work the mic at shorter distances without excessive bass boosting from the proximity effect. Overall, the Baby Bottle makes an exceptionally articulate broadcast microphone.

VOICES IN THE MIX

I next set up shop with fellow engineer Ducky Carlisle at Room 9 from Outer Space studio in Boston. He and I have similar sonic sensibilities, and we put the Baby Bottle through the wringer. We primarily used a Vintech 1272 mic pre, and we recorded both to 2-inch analog tape and to Digidesign Pro Tools through a Troisi DC224ADC A/D converter.

Compared with voices on the radio, which often must stand on their own, vocal tracks to be mixed with music can make for more of a challenge. A lone speaker might sound great through a particular mic, but that same sound might not sit so well in a mix.

Rather than attempt to make a single "do-it-all" mic, Blue intentionally tailored each of its models to specific applications. For example, if you need a bright, airy, naked vocal sound, reach for a Blueberry; if you want a scooped-out sound—big top and bottom but not much midrange—grab a Dragonfly.

What the Baby does on vocals is allow a singer to cut through a bright mix. For example, if you have a great deal of high-end information—cymbals, wispy acoustic guitars, synthesizers, and other sizzle—the Baby will project a vocal right through the middle. Skip the EQ, push up the fader, and the vocal sits right where you need it.

The Baby solved several problems for me on a particularly troublesome vocalist. The singer had a husky baritone voice with some nasal action that fell right within the presence peak of many studio vocal mics. That made it easy to end up capturing too much information. To avoid that problem when recording this singer previously, I generally had used a Neumann tube U 47—a mic that didn't overemphasize his less desirable qualities or nasal sound. The Baby worked especially well in this case: it brought out a smooth midrange without emphasizing the undesirable characteristics of the vo-

calist. The singer still sounded present and forward, but not abrasive or nasal. Moreover, the Baby worked as well going into a Pro Tools setup as it did in the studio through a Manley Laboratories Voxbox (though, admittedly, it sounded a tad better through the Voxbox).

THINGS WITH STRINGS

To test the Baby on acoustic guitar, we had to settle for a Gibson Dove. I typically use small-diaphragm condensers on acoustic guitars (my trusty Neumann KM 140s are the usual pick), though if I'm recording a mono track, I often use a Neumann U 47 or an AKG C 12.

I cut a track of the Dove with the Baby Bottle positioned about six inches from the neck-body joint and angled slightly toward the sound hole. (By the way, positioning this mic is a joy because of its light weight and the lollipop capsule.) The immediately striking thing about the sound was the lack of

low-frequency muck—I didn't hear any of the annoying low-end *whump* that large-body acoustics typically generate. Although the sound wasn't as crisp and detailed as I usually like, it was full, lush, and even somewhat 47-ish.

Interestingly, when I switched to the second of the Baby Bottles, I noticed a bit less bottom end and low-midrange content. Also, the level seemed about 1 dB lower. That suggests that Blue's quality control for the Baby Bottle is not perfect; then again, given the mic's low cost, the quality control is still above average. A 1.5 dB boost of the Trident 80B's low-shelving filter just about matched the sound to that of the first Baby. In the final analysis, I preferred the sound of the second Baby on the acoustic.

On guitar amps, the Baby really sang— or let the source sing, as it were. We dialed up a beautiful flame-top 1960 Les Paul reissue through an original '59 tweed Fender Deluxe. I loved the sound

coming out of the amp, and I stuck the Baby about five inches back from the grille cloth and approximately an inch off the center of the speaker. We ran the microphone through the Vintech 1272 and brought up the fader to hear what sounded an awful lot like the amp in the studio. All of the grit, bark, and nuance of the sound came through. I did hear a bit more ambient sound than I usually expect with a mic at that distance—more evidence of the Baby's wide cardioid pattern—but I had no objections.

For an electric bass track, we set up a '65 Fender P-Bass (strung with flatwounds) through the mother of all bass amps: an Ampeg SVT. We plugged the amp in to a 15-inch Eden cabinet and placed the Baby about three inches from the grille cloth, aimed at the edge of the speaker's dust cover. For a mic pre, we used the Peavey VMP-2, which sounds great on bass. Once again I was pleased with the sound—it was round, full, and perfectly articulate. Granted, the rig sounded great to begin with, but the Baby translated the sound wonderfully without doing anything to alter it.

CRASH AND BANG

Although the Baby Bottle's manual claims the mic "offers numerous advantages when recording drums," I think that needs some qualification. The Baby Bottle is not an overhead mic. When getting drum sounds, I start with the over-

heads. My philosophy is that if the drums sound good to begin with, then well-placed overheads should also sound good; from there I can determine what other mics I need to fill out the sound.

When we tried the Babys as overheads, they threw a tantrum and sounded a bit plastic. They didn't reproduce the depth, dimension, or sizzle of the kit. They didn't do so well as room mics, either. However, I did put a Baby Bottle in front of a kick drum, and the mic handled the sound-pressure level with aplomb, producing a clean, line-level signal. Indeed, it gave me an accurate representation of a lousy-sounding bass drum.

The Baby Bottle did do lovely things with percussion. I usually use mellow-sounding mics when recording percussion, so the Baby played right into my tastes. It was detailed enough on shakers, tambourine, and bongos to sound perfect in the tracks without getting too harsh or irritating.

THINGS WITH MORE STRINGS

I also took the microphones to Rear Window Recording Studio in Brookline, Massachusetts, to introduce them to a Steinway grand piano. This particular piano is on the bright side, and depending upon the nature of the recording, I choose any of a variety of mics (generally Neumann KM 84s or U 47s, Royer SF-12s, or Earthworks TC40Ks). I placed the Baby Bottles in

a standard stereo configuration: one over the treble hammers, the other over the bass strings, and each about six inches above the soundboard. A rich, full, and cohesive sound leaped from the speakers.

Just for reference, I placed a KM 84 next to the treble-side Baby and was surprised by how thin and distant it sounded. Just the same, in a dense pop-rock mix, I would likely steer away from the Babys, simply because they convey too much low-mid information. However, in a sparser tune in which the piano takes center stage, I wouldn't hesitate to use the Baby Bottles.

BACK TO SCHOOL

My time with the Baby Bottles reminded me of several lessons my former Berklee students never tired of hearing. First, all great sounds start with great-sounding instruments and players. Second, when you have a great sound source, choose the mic wisely to convey what you need; avoid EQ.

Baby Bottle Specifications

Element	externally polarized, DC bias capacitor ("true" condenser)
Diaphragm	1", 6 µm gold- and aluminum-vapor-deposited Mylar
Polar Pattern	cardioid
Frequency Response	20 Hz–20 kHz (±2 dB)
Dynamic Range (amplifier)	128 dB
Sensitivity	33.5 mV/Pa (@ 1 kHz into 1 kΩ)
Signal-to-Noise Ratio	87 dBA
Self-Noise	5.5 dBA
Maximum Sound-Pressure Level	133 dB (for 0.5% THD)
Power	48V phantom
Dimensions	8.74" (L) × 1.77" (D)
Weight	0.77 lb.

Third, don't just listen to the mic on its own—listen to it in context with the other tracks. A great sound on its own might not work in a mix.

The Baby Bottle is a mic that doesn't give you what isn't there. Some mics can dress up a sound source and flatter it. The Dragonfly and the Kiwi will do that, but the Baby Bottle won't. Rather, the Baby will give you a clear, cohesive midrange sound that few other mics can, and it is also able to soften strident sources. (Although I didn't have occasion to test the Baby Bottle on brass and strings, I bet that it would work wonders on those instruments, as well.)

The Baby Bottle fills a niche in the Blue line—or in any mic locker, for that matter—and brings a welcome sound into a price category where I have not formerly heard it. In retrospect, I realized that the situations in which I liked the Baby Bottle most were also situations in which I would commonly use ribbon mics—that is, situations for which I wanted a smooth and natural midrange or simply didn't want excessive high-end sizzle. The Baby Bottle was a bit more focused than my ribbon mics, certainly a lot louder, and perhaps not quite as austere sounding.

For people tired of the hype and flab of many budget microphones, the Blue Baby Bottle is the ticket. It not only affords the budget-conscious recordist opportunities previously unavailable, but it also provides the professional engineer with yet another useful weapon in the battle for cool sounds. Whether plugged in to a classic Vintech mic preamp or a Digidesign Digi 001 interface, the Baby Bottle sounds clear and natural. It's definitely the coolest-looking mic in its price class. This is one Baby that won't keep you up by crying and screaming all night. It will, however, keep you up recording all those tracks you've been aching to hear.

Sean Carberry (www.carpedonut.com) is technical director of The Connection *on NPR and a freelance engineer and guitarist in the Boston area. Thanks to Ducky Carlisle for his assistance.*

ST AUDIO
DSP2000 C-PORT (WIN)

Big value in a 10-channel, 24-bit PC audio interface.

By Thomas Wells

The DSP2000 C-Port recording system is a combination of ST Audio's Audio DSP24 PCI card, XG-DB1 daughterboard, and ADC&DAC-2000 converter box. The Audio DSP24 card uses the IC Ensemble Envy24 I/O digital signal-processing chip and supports ten channels (eight analog and two digital) of full-duplex 24-bit, 96 kHz recording. Four DSP24 cards can be linked with sample-level accuracy for a total of 40 channels.

The DSP24 card provides ⅛-inch stereo line inputs and outputs, an ⅛-inch mic input, and two internal aux/CD inputs. S/PDIF optical and coaxial stereo I/O and AES/EBU I/O are provided through a daughterboard that connects to the card with a small cable and mounts in a PC's case slot without plugging in to the PCI bus. The AES/EBU connections use ⅛-inch TRS minijacks; no adapters to XLR connectors are supplied.

A 44-pin connector on the DSP24 card provides an interface that can accommodate a number of specialty external
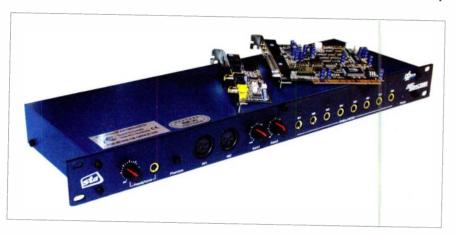
ST Audio boxes to provide multichannel A/D/A conversion and ADAT and TDIF inputs and outputs. The ADC&DAC2000 converter box connects through this port.

The ADC&DAC2000 is an attractively finished, dark blue brushed-metal 1U rackmount box that houses the unit's AKM 4524 converters, keeping them away from the electrically noisy environment inside the computer case. The unit provides eight unbalanced ¼-inch inputs and outputs, a ¼-inch headphone output with volume control, and a single stereo input/output pair with XLR3 connectors. The XLR3 inputs feature gain controls as well as switchable 48V phantom power. The ADC&DAC2000 also provides one MIDI In and two MIDI Outs.

The 6-foot cable that carries data from the PCI card to the box and supplies power from the CPU is long enough for most applications. I did not see a longer cable offered as an option, should it be required.

The outputs of the ADC&DAC2000 are referenced to −10 dBu rather than the +4 dBu professional standard, which means the unit will be headroom-shy in a professional-equipment environment. That and the unbalanced outputs are doubtless the results of efforts to keep

ST Audio's DSP2000 C-Port audio interface is a modular system consisting of the Audio DSP24 card, XG-DB1 daughterboard, and ADC&DAC2000 converter unit. The system offers an assortment of digital and analog I/O, one MIDI In, and two MIDI Outs.

the price of the system low. On the other hand, ST Audio does make costlier external converter boxes that have balanced inputs and outputs.

SOFTWARE CONTROL

The DSP2000 C-Port's software was easy to install on my Windows ME system, a Celeron processor with 512 MB of RAM. Windows driver software is included in the package, and third-party drivers are available for Linux and BeOS.

Signal flow, driver settings, and other hardware settings are controlled by an included application called External Links. **Fig. 1** shows the Main External Links window, a virtual patch bay, where the right-to-left signal routing is managed with colored patch cords.

The External Links patch bay is divided into three main sections. Available inputs are shown at the window's right. In Fig. 1, the metallic-gray rectangle represents the ADC&DAC2000 external box with its four pairs of audio

and MIDI inputs. The ADC&DAC2000 box was chosen from External Links' Add Inputs menu. Immediately below the ADC&DAC2000, the XG daughterboard appears as a choice for S/PDIF or AES/EBU input.

The middle section of the window (in green) shows the DSP24 card's input/output resources. The gray connectors on the right represent the card's input channels, and the white connectors on the left represent outputs. The gray panel marked DAC at the far left of the screen is where "virtual" outputs are routed to the card's actual output channels.

The labeling of the various virtual jacks on the patch bay is not intuitive. At first I was confused by the screen, especially by the middle section of the interface.

In addition to the External Links Patcher, the control software includes an Internal Mixer that manages settings for routing signals to and from jacks and

internal inputs on the DSP24 PCI card. A software External Mixer lets you control settings on the ADC&DAC2000 box.

With all tracks active, the External Mixer application takes up too much screen area, but it cannot be resized. The meters are a bit sluggish, making monitoring difficult. Both mixers can be started from the External Links file menu.

The DSP2000 C-Port system feature I enjoyed most was the ease with which I was able to control signal monitoring. It's possible to monitor each stereo channel pair directly and without any latency by connecting inputs on the middle section of the virtual mixer to outputs on the left section. You might use that, for example, to send out monitor signals of a microphone connected to Input 1/2 to the headphones of a performer (connected to the headphone output of the external box).

DESIGNATED DRIVER

The DSP2000 C-Port distinguishes itself greatly in its ability to route audio to different drivers simultaneously. The unit supports MME, ASIO 2.0, and GSIF (Giga), and it was simple to run Tascam GigaStudio with its GSIF drivers simultaneously with Steinberg Nuendo and Emagic Logic Audio running ASIO 2.0 drivers. You assign drivers in the External Links Hardware Settings window. La-

FIG. 1: The External Links Patcher (above) is used to assign I/O routings across all components. The signal flows from right to left.

tency in the system can be as low as 2 ms.

I tested the unit with a variety of Windows software, and in general, the unit performed beautifully. There were some initial problems with the computer locking up when I used GigaStudio 2.0, but the trouble vanished when I upgraded to version 2.5. The DSP2000 C-Port system is bundled with Emagic Logic SoundTrack 24. It's preconfigured for the C-Port and capable of 16 tracks of audio at a resolution of 24-bit, 96 kHz.

ST Audio's printed manual is adequate, though the section on software control leaves a lot to the imagination. The ST Audio Web site, however, is another story. Besides offering tech support, the site is full of useful information and includes FAQs, a knowledge base, and documentation of users' experiences with the unit. For the most part, the experiences appear to be good, though, to its credit, ST Audio is not afraid to publish comments from disgruntled customers. I was pleasantly surprised at the responsiveness of the tech people: turnaround times were quick, and the techs knew their stuff.

The C-Port is certainly one of the best digital-audio interface values today. The unit is nicely expandable to include options such as ADAT and T/DIF I/O and balanced inputs and outputs. Sound quality is superb: quiet, with clear highs and quick transients that are especially noticeable and appreciated in high-end drum samples. The unit is selling like hotcakes in Europe, and I wouldn't be surprised to see this product quickly become one of the most popular audio systems in the Windows music world.

Thomas Wells has been involved in computer music for more than 25 years. He teaches at Ohio State University.

DSP2000 C-Port Specifications

ADC&DAC2000 External Converter Unit

Analog Inputs	(8) unbalanced ¼" (−10 dBu); (2) XLR3 (−24 to +50 dB) with switchable phantom power (48V)
Analog Outputs	(8) unbalanced ¼" (−10 dBu); (2) balanced ¼" (−10 dBu)
A/D Converters	24-bit, 64× oversampling
D/A Converters	24-bit, 128× oversampling
Sampling Rates	22.05–96.00 kHz
Other Ports	(1) MIDI In; (2) MIDI Out; (1) ¼" TRS stereo headphone
Dimensions	1U × 5.75" (D)
Weight	3.3 lb.

DSP24 PCI Host Card with XG-DB1 Daughterboard

Analog Line Inputs	(1) unbalanced ⅛" (−10 dBu)
Analog Mic Inputs	(1) unbalanced ⅛" (−24 to +50 dB)
Analog Line Outputs	(1) unbalanced ⅛" (−10 dBu)
Digital I/O	(1) optical in (S/PDIF); (1) RCA coaxial in (S/PDIF); (1) optical out (S/PDIF); (1) RCA coaxial out (S/PDIF); (1) AES/EBU in ⅛"; (1) AES/EBU out ⅛"
A/D Converters	24-bit, 64× oversampling
D/A Converters	24-bit, 128× oversampling
Sampling Rates	22.05–96.00 kHz

I M S

INTELLISCORE POLYPHONIC 4.0 (WIN)

Convert polyphonic audio to MIDI.

By Steve Gotler

Are you tired of trying to pick out the notes in songs you want to cover? You can purchase software to play audio files e-x-t-r-e-m-e-l-y s-l-o-w-l-y, but that takes forever, and quite frequently you still can't hear all of the notes. IntelliScore might be able to help. Unlike the audio-to-MIDI converters that are included with some sequencers, IntelliScore claims to convert *polyphonic* audio to MIDI, which is what most musicians really need.

IntelliScore is able to read mono or stereo WAV or MP3 files with sampling rates of 11.025, 22.05, or 44.1 kHz and 8-bit or 16-bit resolution. In addition to converting the notes themselves, the program detects the key of a piece and displays chord names (written as markers) above the notes in the MIDI file. IntelliScore can recognize note durations from whole notes to 128th notes (including triplets) and converts the audio into a format 0 Standard MIDI File. The program installs quickly and without trouble.

FACE FORWARD

IntelliScore's user interface has all of the necessary controls, but many of its elements are a bit crude and inefficient by today's standards. That's a shame, because some rather sophisticated programming lurks just beneath the program's surface.

The most commonly used controls reside in the New Project Wizard and the Project Editor. The New Project Wizard (see **Fig. 1**) helps you set up and complete a project: you select an audio source, set important controls, and generate a MIDI file. The wizard is context sensitive, so it determines what you need to see based on previous choices and displays only that information. That's nice, but its capabilities are limited. For example, the Record Wave File page (see **Fig. 2**) can record

an audio clip from a music CD, but to optimize the audio for later recognition, you need to look elsewhere for better tools.

As you work your way through the New Project Wizard, you adjust several recognition parameters by selecting different settings or by changing values. For example, you can specify whether the audio file is of a single instrument or a group, the probable polyphony level, and the General MIDI playback sound for the MIDI file. You must also indicate the tempo by tapping on the Spacebar as the music plays, or you can allow IntelliScore to guess with its Auto Detect option. After specifying the note value represented by each tap, you complete the conversion process.

Once you have set up a project, you can adjust additional settings in the Project Editor, a multitab dialog box that distributes its controls across five pages (see **Fig. 3**). With the Project Editor, you can fine-tune the recognition process to produce better results. For example, you can specify a more limited note range or adjust the timing sensitivity. IntelliScore lets you save different project settings to make comparisons easier.

However, every time you want to adjust settings (which you do a lot), you have to move from tab to tab, fiddle with awkward controls, acknowledge that it is in fact okay to overwrite the previous MIDI file, and launch an external MIDI player to hear the results. Combine that with the lack of an Undo command, and you can spend a great deal of time trying to convert even a short clip.

IntelliScore could be a lot less cumbersome if Innovative Music Systems (IMS) redesigned the user interface

FIG. 1: IntelliScore's New Project Wizard walks you through a series of steps to convert different audio sources into MIDI.

to improve the controls and to present those controls on fewer tabs, provided built-in audio and MIDI players (the program only provides links to your preferred audio editor and MIDI sequencer), and added multiple levels of Undo. That set of changes would permit you to quickly compare your source audio to the resulting MIDI file and home in on the correct settings before generating a MIDI file for editing.

IN THE MODE

IntelliScore can operate in two modes. In Mode 1, you create a MIDI file by recording an instrument or your voice through a microphone, by recording from a CD, or by selecting a prerecorded audio file.

Most users will probably record pieces from CDs or select prerecorded audio files. The New Project Wizard includes buttons to launch an external CD player and a volume control panel. It also has a no-frills recorder with a meter that helps you bring the input volume into the recommended range. The meter is yellow when the audio level is too low, green when it's within range, and red when it peaks. That's a nice visual aid, but it doesn't make up for the otherwise limited functionality. You'll probably get better results using third-party software. Dynamics compression and EQ may help.

Except with the simplest audio (single-instrument, monophonic) you're un-
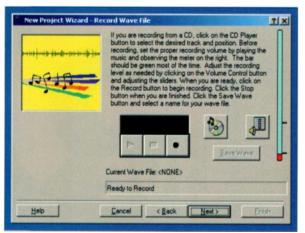
FIG. 2: The no-frills audio recorder lets you record audio from a music CD.

likely to get great results from the New Project Wizard. For the best outcome, you must experiment with the full set of controls in the Project Editor. That can produce dramatically different results and conversion times. On my Pentium III/500 MHz with 256 MB of RAM, some settings increased the conversion time to more than five times the length of the audio clip. It's hard to predict how changing a particular setting will affect the resulting MIDI file, so you could waste a lot of time trying to convert long clips.

The documentation (printed and online) offers some suggestions on how to work more efficiently, such as dividing songs into short clips. I'd take that advice a step further and suggest that you convert only the few passages where you need the most help and work on just a few seconds of audio at a time. In addition, tapping in the tempo may work better than using the built-in Auto Detect feature.

Mode 2 is called Live Performance mode. In that mode, you control a MIDI instrument in real time by singing or playing a non-MIDI instrument into a microphone. For example, you can make a synthesizer follow your vocals. That's a neat feature that performance artists could use. In addition, it is possible to record the MIDI output into your sequencer by installing Hubi's MIDI Loopback Device, a bundled third-party driver that works only in Windows 95, 98, and ME. (IMS recommends the MIDI Yoke driver for use with Windows NT, 2000, and XP.)

MIXED RESULTS

I converted audio to MIDI using both modes. To test Mode 1, I con-
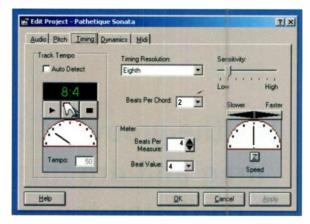
FIG. 3: The Project Editor has all the controls you need to convert audio to MIDI.

verted a few clips from different musical styles; for Live Performance mode, I sang into a microphone.

Included with IntelliScore is a short MP3 excerpt from the second (slow) movement of Beethoven's *Pathetique* sonata. The music consists of a simple melody with accompaniment, is played in a clear and somewhat rigid manner, and was recorded with little or no processing. Although the results weren't perfect, the program correctly recognized the key signature and produced a fairly good MIDI file, even showing the harmonies through chord symbols.

To test the program under more demanding conditions, I converted three other clips: strummed guitars, a piano solo with arpeggios and chords, and a soft-jazz intro with multiple instruments, pitch bends, and percussion.

The strummed guitars produced almost nothing useful. I didn't expect much because of the complexity of the sound, but I wanted to try it anyway. The piano solo and soft-jazz results were about the same. IntelliScore gave me some useable results; I could hear correct notes among the wrong ones, which would have helped me sequence both songs. I e-mailed the piano solo to IMS tech support. It responded within a day with a better MIDI file and corresponding project settings. In Mode 2, IntelliScore tracked my voice fairly well, but it broke up sustained notes into shorter ones and added some extraneous notes.

RIGHT OR WRONG

In the final analysis, IntelliScore Polyphonic proves to be something of an interesting tool rather than a complete solution. According to the documentation, the program recognizes most of the notes in polyphonic audio, leaving you to simply "clean up" the resulting MIDI file. That may be true for some relatively undemanding, highly metrical, clearly recorded music, such as the examples that come with the software, but the audio files that I worked with left me with much more than a simple cleanup job—although IntelliScore did pick out a good number of the right notes, it also "found" many wrong ones.

If the only reason you use IntelliScore is to help you find a few hidden notes in very short audio clips, you probably won't be disappointed. Just don't expect it to generate perfect MIDI sequences. Although the interface doesn't let you work as efficiently as it could, no other program that I'm aware of attempts to convert polyphony to MIDI at all. However, IntelliScore requires patience, decent recording software, and a fast computer to give the best results.

The printed documentation, which is a startup guide, is poorly organized, is poorly written, and lacks important details. The online help fills in much of the missing information, but you may have to hunt for it. The tech support was good, and the response times were fast. Still, with IntelliScore's awkward interface and less-than-spectacular results, it's difficult to say whether it will save you a lot of time in the long run. Download the free demo version at the IMS Web site and give it a spin to help you decide.

By day Steve Gotler is a programmer, a tech writer, and an attorney (www.gotler.com). By night he's gotlerTech, a MIDI-based synth act (www.mp3.com/gotlertech).

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within the an arcing vertical row of LEDs to the left of each knob lights. In the Tube Echo model, for example, Tweak controls the amount of tape wow and flutter that is being modeled and Tweez controls the amount of tube overdrive. In the Digital Delay algorithm, Tweak and Tweez control bass and treble, respectively. The combination of the various delay algorithms and these two controls allows a wide range of interesting effects to be dialed in quickly.

Finally, a Mix knob provides control of the wet/dry mix, and the Power button turns the unit on and off. A minor quibble: the throw of the power button is shallow, making it possible to inadvertently switch the unit off while adjusting the mix level.

SOLID BACKING

A glance at the Echo Pro's rear panel (see Fig. 1) makes it clear that Line 6 is serious about positioning this unit for the professional market—the unit provides stereo ins and outs on balanced +4 dBu XLR jacks and unbalanced –10 dBV ¼-inch TRS jacks. Plugging
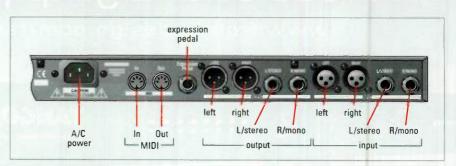
FIG. 1: The Echo Pro's rear panel provides balanced and unbalanced analog I/O (on XLR and ¼-inch jacks, respectively), MIDI In and Out, and a ¼-inch jack for an expression pedal. Conspicuously absent is digital I/O.

TRS cables in to the left side alone lets the unit work as a stereo effects send for other Line 6 products such as the Pod Pro and the Flextone II.

The rear panel also provides an IEC power jack (no wall wart here), MIDI In and Out, and a ¼-inch expression-pedal control-voltage input, allowing more real-time control from a pedal. My biggest gripe about the Echo Pro is the conspicuous absence of digital I/O. AES/EBU in and out would have sealed the deal on this unit as a high-end studio piece, but even S/PDIF would have been welcome.

FORWARD TO THE PAST

The Echo Pro provides 15 delay algorithms, including models of 7 vintage devices (see the table "Seven Deadly Simulations") and 8 Line 6 originals. In ordinary delay applications, each of the Echo Pro's models is very musical. Rather than perfect replicas, however, the simulations can more accurately be described as *reminiscent* of the real things; they capture the general vibe of the originals, but they tend to be cleaner sounding and more mannered. They are devoid of the messy, noisy, unruly qualities that characterize some vintage units—for instance, the Electro-Harmonix Deluxe Memory Man and Maestro EP-1 Echoplex (qualities that are particularly noticeable when you push those units' parameters to their limits, causing circles of feedback with screaming, grungy modulations). Then again, by never quite letting the system go completely haywire, the Echo Pro kind of saves you from yourself.

The Tube Echo simulation sounds like an early tape-loop delay unit. The echoes round off the high end and get distorted somewhat quickly. As mentioned, you can adjust the amount of tape wow and flutter as well as the amount of tube distortion.

The Multi-Head model simulates a 4-head Roland RE-101 Space Echo, creating warm multitap delays. The Lo Res delay model allows you to simulate an early, low-resolution digital delay. However, it is not gritty or harsh enough for my tastes. The Ping-Pong delay model, on the other hand, creates a beautiful, wide delay with the

Echo Pro Specifications

Analog Inputs	(2) balanced +4 dBu XLR; (2) unbalanced –10 dBV ¼"
Analog Outputs	(2) balanced +4 dBu XLR; (2) unbalanced –10 dBV ¼"
Digital I/O	none
Additional Ports	MIDI In and Out; (1) ¼" expression pedal; IEC power jack
Frequency Response	20 Hz–20 kHz (± 0.5 dB; XLR connectors, digital bypass enabled); 20 Hz–20 kHz (± 0.03 dB; XLR connectors, analog bypass enabled)
Signal-to-Noise Ratio	105 dBA
Programs	100 + loop sampler
Delay Algorithms	15
A/D/A Conversion	24 bit
Sampling Rate	46.8 kHz
Internal Digital Signal Processing	24 bit
Delay Memory	24 bit × 64 megabits
Delay Length	2.57 sec. in stereo delay programs
Loop Sampler Length	59.94 sec. with 800 ms echo
Dimensions	1U × 5.375" (D)
Weight	3.44 lb.

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

images bouncing back and forth across the stereo soundscape.

Interestingly, one of my favorite Echo Pro patches is Sweep Echo, an original from Line 6. This algorithm, which adds resonant filter sweeps to the EP-1 emulation, is sure to be a hit with the techno crowd.

The Echo Platter is yet another cool model, based on a strange magnetic-platter echo unit called the Binson Echorec. It creates wobbly, grungy echoes that can get pretty bizarre with all the controls turned up full. The Reverse model grabs a snippet of audio and plays it backward—perfect for psychedelic guitar solos.

GETTING LOOPY

Loop Sampler mode, the 16th setting on the Model Select knob, transforms the Echo Pro into a flexible and full-featured performance looper with 60 seconds of loop time (or a full two minutes when Half Speed is selected). It takes a bit of practice to get the hang of, but once you do, the Loop Sampler is a great deal of fun.

Four of the buttons previously mentioned do double duty as controls for the loop sampler; these are labeled Record/Overdub, Play/Stop, Play Once, and Half Speed/Reverse. You start a loop by pressing the Record/Overdub button. When you are finished recording, you can do one of three things: push Play/Stop, which promptly plays back the loop repeatedly without adding to or changing it; push Play Once, which plays the loop back a single time and then stops; or push Record/Overdub a second time, which puts you in overdub mode, allowing you to layer additional material into the loop.

The fourth dual-function button is labeled Half Speed/Reverse. Tapping it once causes the loop to play back at half speed (and thus an octave down). A double tap causes the loop to play backward. You can even use both functions at the same time. Moreover, you can overdub material while the loop is playing backward (or at half speed) and then play it forward again with the new material now playing backward (or twice as fast).

An 800 ms delay is also available while you are in the looper. This delay affects only the layer that is currently being recorded. Therefore, you can record, for example, a backing rhythm with a bit of slapback and then overdub a lead line with a long delay, followed by a percussive element with no delay whatsoever. Great stuff!

Naturally, getting the Loop Sampler to work well requires good timing on the part of the user. This is a device that makes music in the moment—there are no edit buttons, no undos, and no way to trim the edges of the loop. However, all looper controls can be assigned to Program Change commands, MIDI continuous controllers, or MIDI note values, which means they can be triggered from a keyboard instead of from the front panel.

FREEDOM OF EXPRESSION

The Echo Pro features comprehensive MIDI implementation, allowing the unit to be controlled through sequencers or external MIDI controllers. A MIDI mapper permits translation of any Program Change number to the 99 programs available on the Echo Pro. In addition, virtually every delay parameter can be controlled through continuous-controller commands. Cooler still, all the knobs on the Echo Pro send out those same controller values. That means you can record delay performances into your sequencer from the front panel of the Echo Pro, tweak them in the sequencer, and have the delays respond accordingly on MIDI playback. Echo Pro also responds to MIDI Clock, letting your delay tempos be set directly from the sequence.

The expression pedal is a real-time delay-morphing control. It allows you to shift between various parameter settings within a given delay model. To set the pedal, you must first move it to the heel position and adjust the settings on Echo Pro. Then you have to move it to the toe position and change the parameters again. Afterward, as you rock the pedal back and forth, it changes the parameters in real time, morphing from the first set to the second. That can make for interesting timbral shifts.

Seven Deadly Simulations

Analog Echo	Boss DM-2
Analog Echo with Mod	Electro-Harmonix Deluxe Memory Man
Dynamic Delay	TC Electronic 2290 Dynamic Digital Delay
Echo Platter	Binson Echorec
Multi-Head	Roland RE-101 Space Echo
Tape Echo	Maestro EP-3
Tube Echo	Maestro EP-1

END OF THE TUNNEL

Although I tend to view vintage simulations with a skeptical eye, Line 6 has made me see the light. The Echo Pro has just about everything I could ask for in a digital delay: excellent sound quality, a robust and wide-ranging feature set, and a great, easy-to-grok, real-time, performance-oriented user interface. The loop sampler is easy to use and provides a full minute of sampling time. Dialing in an interesting delay sound is quick and easy. The unit even looks cool.

My only complaints are the lack of digital I/O and the fact that the delay algorithms are not perfect reproductions of the analog effects they are based on (although, admittedly, not all of them claim to be). Just the same, the algorithms all sound very musical, and ultimately, that's what counts. Surely, this terrific-sounding, easy-to-use unit will find its way into many studios and performance rigs. Line 6 has hit one out of the park with the Echo Pro. ●

E - M U

MP-7 AND XL-7 COMMAND STATIONS

They blur the line between live instrument and portable studio.

By Jeff Burger

Dance-music genres have inspired musical-instrument manufacturers to develop one-stop solutions for making music. Two new products from E-mu, the MP-7 and XL-7 Command Stations, integrate sampled pitched and percussive sound sources, a subtractive-synth engine, a multitrack sequencer, a multichannel arpeggiator, touch-sensitive rubber trigger pads, assignable knobs, and a ribbon controller to yield formidable all-in-one workstations.

I reviewed the two Command Stations in tandem because the only difference between them is their sound sets. The MP-7 is targeted at urban modes, and the XL-7 at electronic dance.

CATCH SOMEONE'S EYE

The first thing you'll notice about the Command Stations is that the MP-7 is done up in purples and the XL-7 is bright yellow (see Figs. 1 and 2). Both are tabletop models with bold graphics splashed across the front panel. You can remove the molded side panels to facilitate rackmounting.

In addition to a copious number of knobs and switches, the front panel hosts a headphone jack and a BNC connector for a 12 VDC gooseneck lamp. Connections for MIDI In, two MIDI Outs (for 32 addressable channels), USB, S/PDIF, and E-mu's standard array of three ¼-inch TRS stereo pairs (one master and two subs) are on the rear panel (see Fig. 3). Rear-panel connections are housed under either unit's protruding back lip in such a way that when the unit is rackmounted, it can be a tight fit to insert ¼-inch plugs.

SOUNDS FAMILIAR

In many ways, the MP-7 and XL-7 are evolutionary rather than revolutionary. They're built around the same synth engine as the Proteus 2000 and many of E-mu's other derivative modules. The MP-7 features the sound set from the Mo'Phatt ROM; the XL-7 incorporates the Xtreme Lead-1 sound ROM. Both machines have three additional sockets for additional 16 MB or 32 MB sound ROMs from E-mu, which include Techno Synth, Sounds of the ZR, Protozoa, Orchestral Sessions, Definitive B3, and Siedlaczek Orchestra. E-mu E4 owners can also burn custom sound sets onto flash SIMMs and insert them into the sockets.

For anyone who's not familiar with

FIG. 1: Designed for music production and for live performance, the MP-7 combines the Proteus 2000 synth engine with the waveform ROM and Presets from E-mu's Mo'Phatt Urban Dance Synth.

the Proteus 2000 engine, here's a brief overview (see the September 1999 issue for a review). Samples based in ROM are processed through a formidable 128-note polyphonic subtractive-synth engine that includes a filter, an amplifier, three envelopes, and two LFOs. E-mu's proprietary Z-Plane filter offers 50 filter types ranging from 2nd to 12th order and can morph between two filter setups during the course of an event. Possibilities beyond the standard fare of lowpass filtering include highpass, bandpass, swept EQ, and formants. Matrix modulation lets you connect as many as 24 virtual patch cords from any of 37 modulation sources to any of 41 destinations.

The Proteus 2000 architecture features two effects engines, one offering 44 reverbs and reverb-and-delay combinations and the other serving up 32 assorted effects, including delays, flange, chorus, and distortion. Four buses are available for routing the effects. The dual functionality of the submix outputs extends the effects buses to external gear. The Proteus engine also provides 12 user-definable alternate tunings and extensive MIDI implementation.

E-mu has utilized the multichannel arpeggiator technology pioneered in the Audity 2000 (see the November 1998 issue for a review). The Command Stations can play 32 separate arpeggios simultaneously, each on a different channel and with a different sound. The arpeggiators run through

FIG. 2: Functionally identical to the MP-7, the colorful XL-7 Command Station is oriented more toward techno and electronica musical styles.

the notes of a held chord in Standard mode or trigger preprogrammed note patterns that serve as minisequences. Triggers can reside in sequencer Patterns, or you can assign them to front-panel buttons for live performance. Each unit contains 300 factory arpeggiator patterns and 100 user locations. All arpeggiators sync to the same internal or external master clock as the sequencer, yielding some pretty wild possibilities.

MP-7 and XL-7 Presets can contain as many as four layers that employ the full architecture as well as links to two other Presets. With an additional 12 patch cords per Preset, the potential result is a massive 12-layer sound including keyboard or Velocity splits. You can route Presets to the main stereo outputs or one of the two submix pairs.

Stock units offer 512 factory Presets and 512 user-programmable Preset locations. The number of ROM (factory) Presets increases dramatically with the addition of other sound sets, because each comes with its own Presets. A 32 MB card has 512 factory Presets, for example.

MISSION CONTROL

Although the synth engine isn't new, the inclusion of the control surface and a sequencer take it to greater heights. While you can connect an external MIDI controller to enter notes into the MP-7 and XL-7, you can also use one octave of rubber pads (each approximately one-inch square) for basic input. The Pressure- and Velocity-sensitive pads provide an ideal vehicle for auditioning and entering drum sounds. Transposition switches place the 13 pads anywhere within a seven-octave range. An adjacent ribbon

MP-7 and XL-7 Specifications

Sound Engine	sample playback
Data Encoding	44.1 kHz, 16-bit linear; 24-bit DACs
Real-Time Data Entry	(13) Velocity- and Pressure-sensitive pads; (16) knobs; (2) footswitches; (4) button switches; (16) trigger switches; (1) ribbon controller
Polyphony	128 notes
Multitimbral Parts	(32) internal; (32) MIDI
Presets	(512) RAM; (512) ROM (expandable via SIMMs)
Waveform ROM	32 MB; expandable to 128 MB
Tracks	(16) Pattern tracks with (16) MIDI channels per track; (1) Song track with (16) MIDI channels
Sequencer Patterns	(1,024) RAM
Sequencer Resolution	384 ppqn
Internal Sequencer Storage	300,000 notes maximum
Songs	(512) maximum; SMF import
Arpeggiator	32-channel; (100) RAM patterns; (300) ROM patterns
Effects	(2) 24-bit processors; (60) algorithms
Displays	(1) 2 × 24-character backlit LCD; (1) 4-digit LED
Analog-Audio I/O	(6) unbalanced ¼" TRS outputs; (4) unbalanced ¼" TS aux returns; (1) ¼" stereo headphone output
Digital-Audio I/O	(1) S/PDIF Out
MIDI Ports	(1) In, (2) Out/Thru
Additional Ports	(2) ¼" TS footswitch; (1) 12 VDC gooseneck lamp; (1) USB
Expansion Slots	(3) SIMM sockets
Dimensions	18.60" (W) × 5.25" (H) × 10.50" (D)
Weight	17 lb.

controller called the Touchstrip, though assigned to pitch by default, can serve as a source for any modulation destination.

The functionality of many of the other controls and the information in the LCD depend on the settings in the Mode/View and Edit sections. The Default view shows the transport status, bar and beat, Pattern length, Song number and name, and selected track (for editing and live performance). The Preset view takes you to a standard Proteus-family display for selecting

tracks, channels, and their associated parameters such as volume and pan. The Mix view contains pages with graphics representing volume or pan.

Similarly, a variety of buttons determine what you're editing (Song, Pattern, or Preset) and provide access to pages representing functional

FIG. 3: The Command Station's rear panel has two pairs of assignable outputs and a pair of main outs, one MIDI In and two MIDI Outs, S/PDIF out, two footswitch jacks, and a USB port.

groupings such as controllers or MIDI. Forward and back cursor buttons determine what parameter on the LCD is under the Data knob's control. Dedicated Track/Channel buttons advance and decrement through their respective parameters, providing a welcome shortcut that eliminates the need to move the cursor to a parameter to alter its value.

On the control surface's left side, a grid of 16 knobs serves one of four functions depending on the selected mode. Their default mode is Quick Edit, which offers instant access to common predefined parameters such as filter cutoff, filter resonance, and ADSR segments. Though the knobs are hardwired, Program mode lets you assign them to multiple parameters on multiple channels. The assignments can be stored in any of 63 user-defined Multi setups that might apply to a given Song or performance. In Mix view, the knobs also serve as Volume or Pan controls.

Like the knobs, 16 buttons on the control surface's right side serve various functions depending on the mode. By default, they act as Track Enable and Mute buttons for the 16-track sequencer, allowing easy access for the type of live mixing that is popular with DJ-style performances. Current track-mute settings are saved with Patterns, making it easy to create variations on a theme to use in Song arrangements.

The 16 buttons also serve as event triggers to augment the octave of rubber pads during live performance. You can program each with an associated note, Velocity, and an internal or external MIDI channel. The option to latch an event is great for triggering things such as ambient drones and Presets with arpeggios. You can save the button assignments in Multi setups as you can the knob assignments. When editing Presets, each button corresponds to a

Some things were meant to come in pairs...

GT67
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frequently accessed function page such as the one for the filter or amplifier. As you'll see shortly, the buttons also represent steps when you're sequencing.

LAUNCH SEQUENCE
The 16-track sequencer features Pattern and Song modes. In Pattern mode, each track number has a one-to-one correspondence to the MIDI channel number. The Track/Channel increment and decrement buttons are handy because you can quickly determine which track you're editing or performing. You can also set each track's output to play an internal sound source, an external MIDI source, or both.

While one Pattern or Song plays, you can dial up the next one you want to hear, though you have to hit Enter to lock it in to play at the end of the current passage. Tons of factory Patterns are included, providing plenty of inspiration right out of the box. There's so much quality and diversity that purchasers will likely find themselves initially editing the factory offerings rather than starting from scratch. Factory Patterns are uniformly constructed in that kick is always on track 1, snare on track 2, and so on.

In Pattern mode, you can switch between real-time, step-time, and grid recording and editing on the fly with no break in your creative flow. Similarly, you can switch tracks while recording to build up parts without ever stopping the groove. You can also drop in and out of Record mode while continuing playback, making it easy to audition musical ideas before committing to them.

Grid mode allows programming in the style pioneered on the old Roland TR-808 and TR-909 drum machines. The 16 multifunction buttons represent steps in the sequence. In longer or higher-resolution Patterns, they represent the first block of 16 steps, then switch to represent the next 16, and so forth. You can record multiple sounds on any track, though it's easier to keep things organized if you use a separate track for each sound.

Like grid recording, real-time recording defaults to looping for easy overdubs. Options such as metronome, count off, and quantization make real-time record-

ing a no-brainer. You can also quantize after recording.

Step mode is easy to use, as well. The LCD shows the bar:beat:tick for the current step and provides control over step resolution and gate. At any point, you can switch the autoincrement function on or off, providing the ability to enter chords and Control Change messages.

In addition to establishing a chain of Patterns, Song mode has a multichannel track that runs the entire length of

the Song. It's useful for mixing and for seamless performances that cross Pattern boundaries.

SOFTWARE SUPPORT
The MP-7 and XL-7 ship with a Mac/PC CD-ROM containing E-mu's E-Loader software. With a two-way USB or MIDI connection between the Command Station and a computer, E-Loader lets you retrieve the Pattern and Song lists for easier display. You can also use it to send

and receive the actual Pattern and Song data; that's convenient, as the units don't have any other storage methods. Mysteriously, E-Loader does not do the same for Presets and other information, but E-mu says a fix is forthcoming. Meanwhile, you must rely on SysEx or a librarian to accomplish that task.

E-Loader is also the way to transfer OS updates from E-mu's Web site into the Command Stations. E-Loader has other features, including a graphic grid display of 16-channel MIDI note data received and a text display of all MIDI data received.

The CD-ROM includes a PDF version of the manual, thus providing the best of both worlds. Each unit ships with a 296-page printed manual, but PDFs are handy for searches and for portability.

AT YOUR COMMAND

The MP-7 and XL-7 blur the line between performance instruments and portable recording studios. Some of the technology in either unit is repackaged (you could say *proven*), but E-mu has infused both with goodies that result in powerful, playable groove boxes.

Most pattern-based musicians should find inspiration in the numerous high-quality sounds and Patterns. The Proteus 2000 synth engine is robust enough to keep even the most serious programmer busy for years. The onboard sequencer is powerful, flexible, and easy to master, and the multichannel arpeggiators open new musical vistas. The combination of the rubber pads, Touchstrip, 16 knobs, and 16 buttons is ideal for live performance, whether you're using one of the Command Stations alone or for triggering external gear.

I have few qualms with the MP-7 and XL-7. Almost all that is missing are sampling capabilities and removable storage—features that would add to the price. For the money, E-mu has a pair of winners for the dance crowd on its hands.

Jeff Burger is a songwriter and producer based in Sedona, Arizona.

C - M E X X

MIR

*Another handy weapon
for the control
freak's arsenal.*

By Brian Smithers

For every one of my students who ever cried out in frustration, "Why, oh, why do we have to learn hexadecimal numbers?" I have the best answer yet: the MIR. The MIR is the MIDI Interactive Remote from C-Mexx, and it's a handy little gizmo that uses MIDI messages to control everything from sound modules to effects to computers. Although its ingenious editing application tries hard to smooth the way for the hexadeciphobes out there, you'll be a lot happier and more productive if you are at least conversant with base 16.

The MIR is a hand-size device with three rotary encoders (aka *knobs*) and seven buttons, all of which can be assigned various control functions, such as navigating menus, or MIDI messages of any type (see **Fig. 1**). If you're looking at the picture of the unit and wondering why you see only five of seven buttons, it's because you're not counting the two knobs labeled Enc 1 and Enc 2, which also act as buttons.

I tested the MIR with a couple of available accessories, and I was easily able to get it to control soft synths and digital audio workstations (DAWs). It should be no more difficult to get it to work its magic on hardware boxes, from samplers to mixers. MIDI-controlled lighting would be another perfect match for the MIR—in fact, C-Mexx makes a DMX adapter called Starlight ($545) that expands those possibilities. A wireless adapter is also in the works.

WE HAVE LIFTOFF

The MIR comes with virtually no hard documentation, but it's one of those devices that barely needs it. A 10-foot cable attaches to the MIR through a 15-pin D-sub connector at one end and breaks out into MIDI I/O and power at the other. The power line connects to a wall-wart power adapter. I wish C-Mexx had used female MIDI connectors, because some users will want to place the MIR more than ten feet from their MIDI interfaces, and female-to-male MIDI cables are hard to come by. A new module with female connectors should be available by the time your read this. To the manufacturer's credit, though, the connectors are not the cheap molded type but rather serious screw-apart Neutrik connectors that could easily be replaced with female plugs.

The add-on Live Paq ($195) supplies a pair of momentary footswitches and a variable (volume-style) footpedal. It comes with absolutely no documentation, but again, if you can't figure out that the ¼-inch plugs go in the ¼-inch jacks, you really shouldn't be playing with electricity. I will confess, however, to being irked at the manufacturer for the entire four and a half seconds it took me to realize that the Live Paq cable as-

sembly was designed to replace the original MIR cable, not connect to it. Female MIDI connectors are even more badly missed with the Live Paq because the cable assembly leaves only a foot of separation between the footswitches and the MIDI interface. I had to put the footswitches on my desk until I dug out some female-to-male ¼-inch cables.

The Live Base ($75) is a simple but useful cradle for the MIR that attaches to a mic stand. It doesn't hold the MIR very securely, though a bit of Velcro would fix that. According to the manufacturer, Velcro is now included. The Live Base rotates so that you can set the MIR almost upright or lay it down flat. Left-handers might wish that it allowed the MIR to extend to the left of the mic stand, but that is not possible.

With the hookup taken care of in less than two minutes, it was time to install the software. Because it is a standard MIDI device, the MIR requires no drivers. It does, however, come with an editor, MIR Edit. (As of this writing, only a Windows version is available—you'll need Windows 95, 98, ME, or XP—but a Mac version is in late development.) MIR Edit gives you almost infinite flexibility in programming the MIR's MIDI outputs. I installed the editor from the included CD and then downloaded an update from the manufacturer's Web site. The installation process went smoothly, but note that the editor requires multiclient MIDI-interface drivers to run concurrently with a host application such as a soft synth or a DAW. This meant that on one machine, I had to close the editor before opening a host program. I could easily have worked around that by using a virtual MIDI router such as Hubi's Loopback.

ESCAPE VELOCITY

I fired up one of the factory presets, a set of transport controls for Cakewalk Sonar. Following the excellent step-by-step directions in the online manual (finally, some documentation!), I created a set of "key bindings" in Sonar that would respond to the MIR preset. Presto! I was remotely controlling start, stop, and record; setting and recalling from/to markers; and even using the

FIG. 1: The MIR, the MIDI Interactive Remote, is a handheld remote control for any MIDI-controllable device or program. It features a big bright display, three knobs (two of which function as push buttons), and five buttons.

knobs to scroll back and forth by measure. Any one of the many programs that has similar MIDI message control capability would be as smoothly managed by the MIR.

Turning the knob to the right moved Sonar to an earlier point in time, which seemed backward to me; I opened MIR Edit to remedy that (see Fig. 2). This is where reading hexadecimal comes in handy. Even though MIR Edit has a window that translates the hexadecimal code into plain-language descriptions of the MIDI messages, and even though you can select a new value from a cascading menu of MIDI messages, the Edit window is in hex. Because of that, if you want to change something, you'll be much happier knowing what you're looking at. For novices, the online manual includes a good introduction to MIDI messages and their hexadecimal representations.

I reversed the knob's turn-right and turn-left assignments, saved the updated preset, and uploaded the new version to the MIR. (The upload process is essentially the same as a SysEx bulk dump from MIR Edit to the MIR.) The knob then behaved the way I wanted it to.

Of course, that whetted my appetite for more power, so I set about creating some additional banks of Sonar controls. First, I assigned a new set of MIDI note messages to the MIR's buttons and knobs, and then I created some additional key bindings in Sonar that allowed me to arm the track for recording, punch in and out, set auto-punch locators, and undo a recording.

FIG. 2: MIR Edit allows you to assign any MIDI function to the MIR's buttons and knobs. Controller mappings and aliases expand the possibilities.

I was able to accomplish all of that from across the room, safely away from my computer's noisy fan.

Next, I created a MIR bank consisting of volume and pan controls for 16 Sonar mixer channels. I used the MIR's first encoder to scroll through the controls and the second encoder to adjust the control's value. That let me move away from my computer keyboard and settle into my monitors' sweet spot for optimum listening.

I set the center button to allow me to switch between the control banks I had created. The MIR stores banks in nonvolatile RAM, and I had a difficult time eating up even a fraction of the available space. With some forethought and a little programming time, I was able to bypass my computer keyboard almost completely.

IN ORBIT

Up to this point, everything I needed to know in order to program the MIR

MIR Specifications	
Buttons	(5) buttons; (2) encoders also act as buttons
Knobs	(2) rotary encoders; (1) "analog" pot
Connectors	DB15 multipin; 10' breakout cable with MIDI In and Out (male) and power
Power	7.5V
Display	2 × 16-pixel LCD
Dimensions	6.0" (W) × 4.5" (H) × 1.5" (D)
Weight	0.4 lb.

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was well laid out in the online manual. Unfortunately, some powerful and important aspects of MIR Edit are not so well documented. For example, Mappings allow you to apply various curves to the encoder data. Two logarithmic scales—as well as sine, cosine, square, and linear curves—are available, and you can offset the curve horizontally or reverse it horizontally or vertically.

One of the MIR's attractive qualities is its big, bright display, and you can set up Alias Tables that can be used to display user-friendly names for the various parameters you are tweaking with the MIR. Icons are also available, though some of them are less useful than others. The documentation for Mappings, Alias Tables, and Icons will leave many users bewildered, as it jumps into programming details without giving adequate background on what they do and how they are useful. However, I have recently learned that new documentation, covering these and other features, will be available soon.

The Live Paq would certainly be a useful addition in a live setting, but it was an asset in my studio, as well. I was able to spread my controls out between my desk area and my keyboard area, which saved me a lot of rolling back and forth. Be aware, however, that the Live Paq adds only one new control assignment. Switch 1 duplicates the function of the MIR's center button, and the continuous pedal replaces (and therefore disables) the "analog" pot. Only Switch 2 carries a new independent assignment.

MIR Edit has the kind of depth that will allow motivated users to do some amazing things—from reconfiguring devices with System Exclusive dumps to playing MIDI files stored within MIR's memory—with the little controller, and I expect to see a healthy subculture of MIR bank swapping in time. In fact, new presets for various types of gear appeared on the company's Web site while I was writing this review. Moreover, C-Mexx says several companies—including Cakewalk, Native Instruments, and Steinberg—have recently signed on to implement the MIR protocol. In addition, the recently announced advanced tutorials should go a long way toward making up for the manual's shortcomings.

If you think the MIR might be useful to you, check out MIR Simulator, a fully functioning software demo that is available from www.mircontrol.com. Intended as a way to test your banks prior to uploading them to the MIR or to work on your banks when your MIR is not connected, it's being offered as a great teaser to introduce you to the system's myriad possibilities. A word of warning, though: if you use MIR Simulator as I did to control your DAW's mixer, try not to think too hard about the fact that you're using a software emulation of a hardware control device to control a software emulation of a hardware mixer. It'll only make your head hurt.

Brian Smithers *is a musician, an engineer, an educator, and a writer in Orlando, Florida. He teaches at Full Sail Real World Education in Winter Park.*

VERMONA ENGINEERING
PH-16

By Alex Artaud

. . . .

The PH-16 ($339) is a dual eight-stage phase shifter from Germany's Vermona Engineering. Although the device looks modest, you can use the phase shifters in parallel or serially. The PH-16's host of modulation options gives you a surprisingly large palette of filtering and spatial effects.

Phase One

The first thing you notice about the PH-16 is its array of large silver knobs, which are inviting to those who like to tweak analog gear. The knobs have a nice feel to them, but unfortunately, their shininess makes it difficult to see the notches that indicate the settings.

The 1U front panel is divided into five sections. At the far left is the Input section, which has a Gain knob, a clip indicator, and an unbalanced, high-impedance ¼-inch input. In the next section are independent Phase and Feedback controls for each phaser. Between them is the Mode knob that selects whether the phasers are working in parallel (in 4 , 6 , or 8 stages) or serially (in 8 , 12 , or 16 stages). One drawback of the PH-16 is that you hear a pop when you change modes.

The PH-16 has an internal LFO (with triangular wave only) and controls for Speed and Intensity. A flashing red LED shows the LFO speed. With the Destination knob, you can route the LFO in one of four ways: to Phaser 1 only, to Phaser 2 only, to Phaser 1 and Phaser 2 simultaneously, or to Phaser 1 and Phaser 2 simultaneously but with an inverted modulation waveform for Phaser 2.
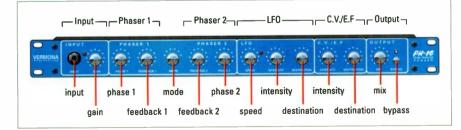
Similarly, the Destination control in the Control Voltage/Envelope Follower section routes the input at the CV/In jack in six ways: in addition to controlling the phasers as previously mentioned, you can also control the LFO speed (SPD) and intensity (INT). When no CV source is plugged in to the rear-panel CV/In jack, the circuit functions as an envelope follower, which is affected by the signal level at the input. The stronger the input, the higher the voltage level going to the selected destination. The final section, Output, has a Mix control and a Bypass button (with LED indicator) that pops when you press it.

The rear panel has two main audio inputs: Phaser 1 and Phaser 2. When you plug an instrument in to the front-panel input, the two rear-panel inputs are defeated. Phaser 1 is an unbalanced ¼-inch input that sends the signal to both phasers. The Phaser 2 input accepts mono and stereo inputs—when a TRS plug is inserted into this input, the Phaser 1 input is defeated, and the two channels of a stereo signal going into Phaser 2 can be processed independently.

Additionally, each phaser has a dedicated unbalanced ¼-inch output. In Serial mode, the Phaser 2 output gives you the combined signal of both processors. Other rear-panel jacks are a bypass footswitch, a control-voltage/pedal input, and the wall-wart power supply. The PH-16 has no power switch.

Sound in Motion

I used the PH-16 in Serial mode for conventional chores, such as tracking guitar and bass lines. On the guitar, it was easy to get results that ranged from familiar phaser whooshes to rapid watery LFO sounds. I could also make bass tones sound rounder or metallic, depending on the settings.

Drum loops sounded great when processed through the PH-16 in Parallel mode. While varying phaser feedback, changing modes, and relying heavily on my expression pedal, I recorded repeated passes until I got the wild variations I was looking for. You can hit the PH-16's input fairly hard to get a pleasant distortion—nice.

Dual phase shifters working in parallel are perfect for creating weird filtering treatments and swirling spatial effects. The PH-16's handcrafted optocouplers are part of the reason this box works as well as it does. Less expensive analog phase shifters tend to be heavy in the midrange, but the PH-16's wider bandwidth made me feel like I could craft the space around any sound.

The PH-16 sounds a bit brighter than other hardware phase shifters I've used. Its smooth sound, however, makes it superior to any phase-shifter plug-ins I have in my collection.

Phased Out

Although the PH-16 looks simple on the outside, it can process audio in surprisingly complex ways. Consequently, you have to change the settings carefully if you want to get the most out of this device. The modulation can become seemingly unpredictable quickly, and if you turn the knobs too far too fast, you may miss something interesting. The PH-16 is a sturdy processor with a unique sound, and it's definitely a hands-on unit, with plenty of knobs that make it fun to use.

Overall EM Rating (1 through 5): 4

Enport (distributor); tel. (402) 398-0198; e-mail info@vermona.com; Web www.vermona.com

Vermona Engineering's PH-16 is a dual eight-stage phase shifter that allows you to use its processors serially or in parallel.

TECH 21
SansAmp XDI

By Barry Cleveland

. . . .

Tech 21 designed the SansAmp XDI ($95) primarily as an interface for its PSA-1 Pro Tools plug-in—the software version of the company's popular PSA-1 guitar amplifier

and speaker cabinet simulator—but it also functions as a standard direct-injection box when used with mixers and computer audio cards.

There are two high-impedance (1 MΩ) ¼-inch inputs labeled Normal and Bright, just as you'd find on a typical tube guitar amp. Unlike a guitar amp, however, you can use only one input jack at a time. A balanced XLR and an unbalanced ¼-inch low-impedance output jack are provided. The latter eliminates the need for an adapter when using the XDI with audio cards that have only unbalanced ¼-inch inputs. You can use both outputs simultaneously, though the manual advises against it with a warning that you could compromise the sound quality of the ¼-inch output.

In the Black

The XDI is housed in cast aluminum painted black with yellow labeling and measures 4.5 inches wide by 3.5 inches high by 1.5 inches deep. The casing feels sturdy, and the unit's four solidly attached audio connectors appear to be of high quality. Inside, the wiring and solder work is neat and clean. A hinged battery door on the bottom plate is a thoughtful addition.

The use of FET technology results in remarkably quiet operation. The circuitry accepts power from a 9V battery, a 9 VDC power supply such as the optional Tech 21 DC2 ($12.95), or 48V phantom power through the XLR output jack. The unit will operate with as little as 24V, making it compatible with many less expensive mixers and preamps. Conveniently, the unit over-

rides the battery when it detects phantom power.

Well Connected

I don't own a Pro Tools system, so I was unable to use the XDI with the PSA-1 plug-in. However, I did use it to drive the amp simulator in my Yamaha 03D digital mixer. The XDI made the mixer's rather lame amp simulator sound more realistic—an unexpected benefit. I patched the XDI between my PRS Custom 24 guitar and the 03D mixer using every possible configuration of inputs and outputs.

First, I connected the XDI to the 03D using the XLR output, powering the box with the mixer's 48V phantom power; then, I connected the XDI using the ¼-inch output, powering it with a 9V battery. Both methods yielded excellent results, and I could not perceive any differences in audio performance.

On the other end of the box, the XDI's Normal and Bright inputs offer distinctly different sounds, both of which will be familiar to guitarists who are used to playing through vintage and vintage-style tube amps. The equalized sound of the Bright input is particularly authentic sounding, providing the sort of transparent high-frequency sheen reminiscent of an old Vox AC30, for instance.

Next, I patched the XDI into one of the ¼-inch inputs on my Mark of the Unicorn 2408mkII audio interface and recorded some guitar sounds directly into Digital Performer. The XDI sounded great in this application, functioning almost as a preamplifier in the way that it tightened up the sound overall and added high-end definition. The unit sounded the same when powered with a 9V battery as it did when powered with an external power supply.

Speaking Directly

There's really little more to say about the XDI. At $95 it's a great bargain for guitarists (or bassists) who want to plug their instruments in to a mixer or sound card, especially if you want your instrument to actually sound like a guitar without adding noise or undesirable distortion. You can

never have too many direct boxes at your disposal, so check it out!

Overall EM Rating (1 through 5): 4.5
Tech 21; tel. (212) 315-1116; e-mail info@tech21nyc.com; Web www.tech21nyc.com

YELLOW TOOLS
Pure Guitars
By Dan Phillips

Although my main instruments are keyboards and percussion, I've always been fascinated by guitars. I recently took one down from the rack at my local music-gear superstore, held it to my body, visualized perfection—and played a poorly fingered E-minor chord. Maybe some day I'll sit down and learn to play; for now Yellow Tools' *Pure Guitars* ($149.95) does a pretty good job of fulfilling my jones for an acoustic guitar.

The collection is available on CD-ROM in Akai/E-mu, Giga, or Emagic EXS24 sampler formats. For my review, I used an E-mu E6400 Ultra sampler with 128 MB of RAM. *Pure Guitars* favors quality over quantity. The collection presents seven extensively multisampled guitars: Nylon Concert, Nylon Spanish, Steel Pick, Steel Fingered, Steel Jumbo, Ovation Piezo, and 12-string. There are also sets of chords from the Ovation and 12-string and two multisamples of acoustic bass.

Take Your Pick

The bulk of the sounds share a refreshingly regular layout; the sound designers were clearly well organized. All of the main programs take up 32 MB and use five levels of Velocity switching: soft, medium, loud, hard attack with string buzz, and a bend up to the root pitch. Most include a separate harmonics program with two-way Velocity switching, along with 60 or 70 FX Samples of string scrapes and body knocks—pretty hefty stuff.
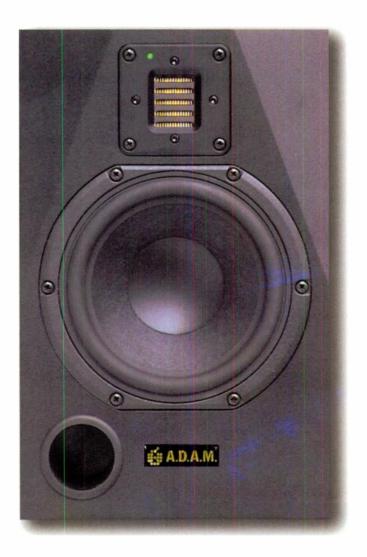
Rather than resorting to loops, Yellow Tools captured the full, natural decay of all of the instruments. Samples are usually mapped to cover a few half steps, keeping pitch-shifting artifacts to a minimum.

Tech 21's SansAmp XDI is a convenient and versatile direct box for interfacing guitars and basses with digital-audio sound cards or mixers.

Mixing on normal monitors is like trying to enjoy a gourmet meal with a head cold.

I magine how great it would be if you could wave a magic wand and suddenly hear *everything* more clearly. Details that used to be buried in the mix would leap out, even the smallest tweaks would be fully audible, you could place things in the sound field perfectly, and all of your gear — all your mics and pre-amps and processors and instruments — would be that much more useful and effective.

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Monitors are monitors...but ADAM Analyticals are more!

The samples are close-miked and in mono, resulting in an intimate but rather dry timbre. For best results, add a touch of reverb.
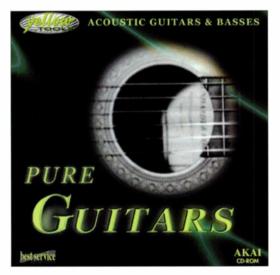
Up for the Down Stroke

Almost all of the programs sound great—they're expressive and fun to play. Steel Jumbo has a sweet, round tone, possessing a steely tone without being overly bright. The nylon programs are open and pure, with Nylon Spanish offering a slightly fuller tone than Nylon Concert. The Ovation Piezo harmonics are really neat, with a timbre that straddles the line between acoustic and electric. The wild, organic character of Aggressive Acoustic Bass would be perfect for rootsy rock. I have other sets of sampled chords—some including distinct sets of samples for up and down strokes—but *Pure Guitars'* 12-string chords are the best I've heard. A three-way Velocity switch adds to their realism.

Still, the collection isn't completely perfect. Several samples have clicks at the end, which usually show up when holding down the sustain pedal. The clicks were easy to edit out, but I wish that they weren't there in the first place. Also, one program had a gap in its complex Velocity map, such that a Velocity of exactly 123 would play no sound at all.

Strum Away

Pure Guitars' great samples, strong organization, and playability overshadow

those few weaknesses, especially because of its reasonable price. I recommend it to anyone who needs to create realistic acoustic-guitar parts with their samplers as well as to keyboardists who just want to sit down and strum every once in a while.

Overall EM Rating (1 through 5): 4
EastWest (distributor); tel. (800) 969-9449 or (310) 271-6969; e-mail info@yellowtools.de; Web www.yellowtools.com

SPECTRASONICS
Backbeat
By Rob Shrock

● ●

Put four world-class drummers into a studio with Eric Persing, and you're bound to catch some magic. *Backbeat* is the second title in the Classic Drumming series from the masters of sound design at Spectrasonics. Offered as a CD-ROM set in Akai/E-mu or Roland formats with Groove Control ($199) and as an audio CD without Groove Control ($99), *Backbeat* focuses on straight-ahead drum grooves.

Super Loops

Backbeat consists of stereo drum loops only. Individual drum hits are not provided, though cymbal and side-stick samples are included to augment the loops. Each of the 39 grooves contains several loops of basic beats and minor variations, as well as several loops with corresponding fills. The tempos of the original loops vary from 65 to 148 bpm. The main grooves with their variations and related elements fill three CDs with more than a gigabyte of material. That provides a great deal of flexibility for mixing and matching, and the *Backbeat* loops all sound spectacular.

Together, Persing and Bob Wilson recorded Gregg Bissonette, John Ferraro, Eric Boseman, and Wilson at four world-class studios in Los Angeles. Little space is wasted on esoteric beats; the emphasis is on maximizing the

Spectrasonics' *Backbeat* sample library offers a wealth of superb, useful drum-set grooves played by Gregg Bissonette, John Ferraro, Bob Wilson, and Eric Boseman.

musical usefulness of every groove. The loops are individually processed with the EQ, compression, and room effects that are appropriate for each style, and they are almost all spot-on. The drums sound full and deep on the bottom end and clear and punchy on the top end. In other words, the rest of your tracks better be up to snuff, or these drums could show you up.

Taking Control

The CD-ROM version of *Backbeat* is in Groove Control format, which means that the loops are sliced up into individual beats that are mapped across several MIDI notes. Standard MIDI Files for each groove are provided to re-create the loop when it's loaded into your sampler and played from a sequencer. The name of each loop indicates the original tempo, but Groove Control lets you change the tempo, feel, and pattern of the loops using the tools and commands in your sequencer. You can make a swing tune out of a straight loop, for example, or turn a funk groove into a speed-metal track.

Because several complete kits were recorded in different studios, you aren't stuck with one "sound" for the entire library—another big plus. The loops consist of great-sounding drum kits, rooms, and processing that make the loops mix-ready. Wisely, no auxiliary percussion instruments such as shaker or tambourine were included in the loops, which could have limited their usefulness.

Yellow Tools' *Pure Guitars* sample library focuses on realistic reproductions of seven acoustic guitars.

The loop styles are geared mostly toward rock, funk, modern country, and progressive jazz. I'm using a combination of "88-Solid Rock" and "91-Push Pop" for a modern-country song demo. Thanks to Groove Control, I have no problem combining the loops at my required 92 bpm.

A fourth CD, a Mac/Windows "data" disc, is a real goody box. All of the loops are provided again in WAV format (essentially the audio version of *Backbeat*), and a custom sound-file player called Sound-Finder MVP is included. SoundFinder MVP enables you to quickly audition the individual WAV loops on the data disc so that you don't have to load the loops from the other discs into your sampler to hear them. In addition to SoundFinder, the data disc includes the accompanying MIDI files, PDF manuals for *Backbeat* and Groove Control (printed documentation is also included), demos that highlight the drum patterns, and a couple of handy utilities.

Get Your Groove On

Overall, I have no complaints about *Backbeat,* except that I wish I had individual hits from the sampling sessions; the drum sounds are truly superb. You can get individual hits from the Groove Control data to build your own kits, but that's only moderately successful, because you must cull the hits from the existing loops. Nevertheless, *Backbeat* is an extremely useful collection of loops, and you'd be hard-pressed to find grooves that feel and sound any better.

Overall EM Rating (1 through 5): 4.5

Ilio Entertainments (distributor); tel. (800) 747-4546 or (818) 707-7222; e-mail ilioinfo@ilio.com; Web www.ilio.com or www.spectrasonics.net

PROPELLERHEAD SOFTWARE

Strings (Mac/Win)

By Rob Shrock

• •

Propellerhead Software's *Strings* ($89) is the company's first offering in the Reason ReFill collection. ReFill titles include instrument samples and REX2 (ReCycle format) data that are designed as add-ons to Propellerhead's popular Reason virtual-

studio environment. The *Strings* sample library contains 288 REX2 phrases for use in Reason's Dr. Rex player and 42 multisampled instruments for Reason's NN19 sampler.

Fine Phrasing

The bulk of *Strings* consists of musical phrases that each last for several measures. They would serve well as backing accompaniment to an existing track or as the centerpiece around which to build a new composition. The phrases are divided into four categories: Orchestral Strings, String Quartet, Upright Bass, and Harp. File names contain musical key and tempo information, such as 120-Amin-StrQrt-All.rx2 or 095-Bmin-1b-OrchPhrases.rx2.

Each string-quartet phrase is provided in an ensemble version as well as in four solo versions that are played on each instrument. Dr. Rex allows you to easily change the tempo and key of the loops, so a little creative editing can milk a lot of mileage out of the phrases.

The quality of the writing and the performances of the REX2 loops are top-notch. Recorded in one of Moscow's finest studios, each phrase possesses a natural ambience that sounds perfect for the various styles. The intonation and tone of the session players are uniformly excellent.

The two ensembles (full string orchestra and string quartet) are premixed to stereo. The balance of the strings is excellent, although the panning of the string quartet is a little wide for my taste. However, it is a simple matter to make the panning a bit narrower if you prefer, or you can play the four solo versions of the quartet and remix them to your liking. Little harp material is included—mainly just a handful of arpeggios—and there is no multisampled harp instrument.

Lite Instruments

Speaking of instruments, the individual sampled instruments are not the strongest part of this collection. You get a few really good patches; however, most of the solo instruments are looped too tightly, which gives them a mechanical feel. When used as add-ons against the backdrop of the looped REX2 phrases, the included multisampled instruments are moderately more

Propellerhead Software's *Strings* sample library, the first in the company's new Reason ReFill series, includes instrument samples and REX2-format files for use in Propellerhead's Reason software.

successful. This collection doesn't stand up as a solid library of string samples, though, especially when compared with other available string-sample libraries.

My biggest criticism is reserved for the library's overall lack of content. The entire disc, including the REX2 files and individual instruments, adds up to little more than 300 MB. The collection of REX2 phrases covers a wide spectrum, and there I have no complaints. As good as the Russian musicians sound on the ensemble REX2 phrases, a more complete collection of string instruments could have been created, which would have better rounded out the library. For whatever reason, the sound designers missed an opportunity to set a higher standard for the Reason ReFill collection.

The concept of ReFill titles for Reason is a good one, and *Strings* is a decent start. The looped phrases are great, and many people will find them immediately useful as a tool for commercial production and as a way to learn more about good string writing. Next time I'd like to see Propellerhead fill up the disc with more of the good stuff! ◉

Overall EM Rating (1 through 5): 3.5

Midiman (distributor); tel. (800) 969-6434 or (626) 445-2842; e-mail sales@midiman.net or info@propellerheads.se; Web www.propellerheads.se

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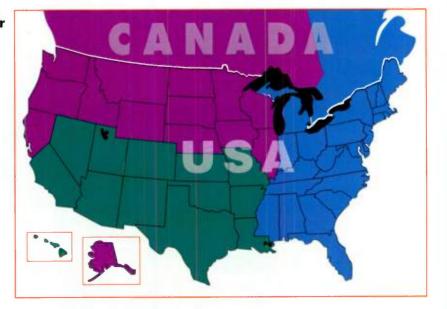
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FINAL MIX

Do People Care?

Do you care what people think of your work? How about what they think of you? Should you care? Why? These are simple questions, but they are surely not easy ones.

If a client hires you and then says he or she doesn't like the job you've done, should that matter to you? If you value that client or your reputation among those people the client speaks to, then I would say it should.

If a band member tells you that you're difficult because you insist your song be played a particular way, does it make any difference to you? What about if a reviewer says your album's no good and your hair is bizarre?

There is no definitive answer to these questions; too much depends on circumstances. For a start, consider the source. Is the person speaking some wanna-be rock-star lead vocalist who wants to build him- or herself up by tearing you down, or is it the person who produced three out of five of your all-time favorite albums?

Another thing for you to consider is how an opinion is offered. Is it a sweeping statement barreling down a highway of vague generalities, or does it sound reasoned and specific?

Logical considerations are just the beginning: are you certain of what you're doing, or are you feeling your way? Are you in a mood to hear what anyone has to say? Each of these factors can be liquid and can land at either extreme of your personal scale of validity—or in the continuum between.

Many years ago, my father—clever man that he is—taught me the proper conjugation of the word *stubborn:* "I am *stubborn,* you are *obstinate,* and he is a *pig-headed fool.*" No less than David Crosby sang, "Hear, you must hear what the people say" ("Long Time Gone"). It is indeed the fool who refuses to ever listen to what anyone tells him or her.

Fine. But is refusing to listen when someone is laying The Word on you inherently foolish? Albert Einstein (who more closely resembles my father than Crosby in that he no longer tours much) once said, "Great spirits have always encountered violent opposition from mediocre minds." Or, as Ray Davies of the Kinks more succinctly put it, "I'm not like everybody else." I'm sure I've been identified with the third-person-singular conjugation in my father's definition at times; perhaps this is why. Or maybe I'm just full of myself.

Now that I've chased in circles for a few hundred words, I'm supposed to come up with a neat and snappy conclusion, and I'm going to (more or less), but I'll bet I get as many flying eggs for it as nodding heads. So be it.

If there is no definition of "solution" and no stability in circumstance, how do you sort out whether to care about what people think of you and what you do?

The only thing I can trust that won't lie to me is my gut feeling. I can fool myself with my brain, and I can lie to myself with my heart, but my gut has proved its accuracy time and again, and In Gut I Trust. That much presents me with a nice, pat solution. The trick, of course, is keeping everything else from obscuring the signals, and that turns out to be quite a chore.

Sorting through my take on the validity of the source, subduing my ego, and avoiding double-thinking myself into fatal insecurity—all those things and more are like the summer fog that envelops San Francisco in the late afternoon when the Central Valley heats up. If I can keep tabs on the temperature in the Central Valley and see the fog coming, I might be able to hightail it to the top of Mount Tamalpais, where I'm above the fog. If not—well, Dad also used to say about me, "often erroneous, never in doubt."

But I'm willing to take that risk because there is more than one right answer to many things in this life, and no one has a lock on The Truth. Or, as the famous philosopher Anon said, "You can't believe everything you hear." ◉

STEVE CURL

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