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June 2000

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S ince we introduced the MX-2424 hard disk recorder, there has been a lot of speculation about its price (which is so low it seems too good to be true).

So we get questions. Like...

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

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

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

"*Does it need an external computer?*" (No. The MX-2424s front panel has a full set of professional transport, editing, and track assignment controls, including a shuttle/ scrub knob. So you don't have to have a computer to run it. But — if you happen to own a Mac or a PC, you can take advantage of the digital audio editing and control software that comes standard with each MX-2424 to do even more. Your choice.)

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

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

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

$3,999.ESP*

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

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

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Electronic Musician

May We Quote You?

To our chagrin, we've recently discovered some of our reviews and quotes from our reviews on manufacturers' Web sites in advance of the articles' publication. Even more disconcerting, the copies posted on the Web weren't the final, completely edited articles. What's going on here?

First of all, I am convinced that nobody was trying to pull a fast one. A few people apparently misunderstood our policy on "sampling" our editorial. In every case, the offending text was withdrawn as soon as we requested it. But why reprint partially edited reviews—and without authorization, at that? For that matter, how do you get permission to reprint our articles? When you read quotes attributed to us, how do you know that they're legitimate?

Of course, we recognize and encourage "fair use" rights to reprint our work for educational purposes, commentary, criticism, parody, news reports, and the like. No permission is required for such reprints. So teachers relax—you can make copies for your class. And quoting us online in order to comment on an article is fine, too.

But a manufacturer that wants to reprint our articles for a marketing campaign has to request permission. If an entire article is going to be reprinted intact—and provided for free to the public—reprint permission is easy to get. (Selling reprints requires permission, too, but that is a different proposition.) We do insist that only the final, published version of the article be used and that it has been in print for at least one month before anyone reprints it.

We guard quotes more closely: the exact quote must be submitted for approval. We make sure that the quote is accurate and that the author's meaning hasn't been warped before we grant reprint permission.

So how did the partially edited articles get on the Web? As soon as an editor completes a product-related article, we e-mail it to the manufacturer for a fact-check. The manufacturer discusses factual miscues and other aspects of the article. We then make the changes that we deem appropriate.

After the fact-check, the article is further edited and proofread several times, so it may change significantly. But the manufacturer won't see those changes until the magazine hits the streets. The partially edited articles we found on the Web were fact-check copies.

We want to let companies use our material, while at the same time preserving the integrity of our editorial content and protecting consumers from misrepresentations of our views. Luckily, these mistakes don't happen often.

• • • • • • • • • • • • • •

Several confused readers have contacted us in search of Sanyiu Samples' *Sounds of Silence* sample CD, which Larry the O reviewed in our April issue. I hesitate to admit this, but the review was an April Fools' joke. We thought the looped digital silence and the mic inside the permafrost would be sure tip-offs, even if you missed the other hints. The sample library's creator was Jeok Sanyiu, and the company's Web URL was www.thejeoksanyiu.com. Translation: the joke's on you! We published three more April Fools' jokes in the same issue; did you find them?

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APRIL FOLLY

I bought the *Sounds of Silence* sample CD reviewed in April [see "Quick Picks: Sanyiu Samples" in the April 2000 issue of **EM**], and I want to mention something that the reviewer ignored: none of the underwater recordings hold any "air."

The joke is in the spirit of **EM**'s forerunner *Polyphony*—nice to know that the tradition holds.

Terry Truhart
via e-mail

Terry—I'm glad you enjoyed it. You wouldn't believe how many people we fooled—so many that I had to fess up to the joke in this month's "Front Page" column.—Steve O.

ON THE WRONG TRACK

As a Macintosh user, I was not "a-Mazed" by your cover story [see "Making Tracks" in the April 2000 issue of **EM**]. It goes on for 37 pages (including ads). I'm interested in making music and recording, and so are many PC users, but 37 pages is too much. Do I have to buy a PC to have an interest in your magazine? Are you telling me that there are no comparable products on other platforms? Is your magazine a subsidiary of Microsoft Enterprises? If you do not have the intellectual subjectivity (because

you use Windows?) to edit a magazine for all your readers, leave the job to someone else.

You went on for 37 pages without considering even one program on any other platform. A big chunk of the magazine was useless to me and many others. Pro Tools, *Cubase, Digital Performer, AudioDesk,* BIAS *Peak,* and many other programs have Mac users. I know my voice doesn't mean much to you, but I am really frustrated by your attitude, and as tiny as my voice may be, I'll be spreading the word from now on!

Daniel Cabana
via e-mail

Daniel—On the contrary, your voice means a lot, which is why I'm replying publicly and at length. To begin with, we write about a variety of Macintosh and cross-platform products. In fact, we reviewed four cross-platform products in the same April 2000 issue in which "Making Tracks" appeared. That doesn't include the lead review, which was the Mac-only Digidesign Digi 001, featuring in-depth coverage of Pro Tools LE 5.0.

"Making Tracks" discussed audio-only multitrack editing programs, not digital audio sequencers such as Mark of the Unicorn Digital Performer and Steinberg Cubase or 2-track programs like BIAS Peak. We covered six dedicated multitrack audio editors for the PC—more than enough for one article. Had I dealt associate editors Dennis Miller and David Rubin one more program, they might have gone insane. (By the way, both Miller and Rubin use Mac and Windows computers for music production.)

We ran a cover story on digital audio sequencers for Mac and Windows ("Sequencing Games") in July 1999 and a cover story on 2-track audio editors for Mac and Windows ("Shaping Better Waveforms") in March 1999. Most of the versions that we wrote about are still current. We also have separately reviewed the current versions of all the products that you named except Digital Performer/AudioDesk, and our upcoming review of Digital Performer 2.7 will include full coverage of its audio features. (AudioDesk is basically the audio section of Digital Performer.)

Versions 5 and higher of Pro Tools *are digital audio sequencers. Otherwise we would have included it in "Making Tracks," because it runs under both Windows NT and the Mac OS. In the July 2000 issue, we will present a huge review of the Pro Tools/24 Mix system and the full-featured* Pro Tools 5 *software.*

There remain only two dedicated multitrack audio editors for the Macintosh: AudioDesk, which I have discussed, and BIAS Deck, which hasn't had a major upgrade in several years. We will review Deck when a significant upgrade is released. So eventually you will get everything you asked for.—Steve O.

THE WHOLE TRUTH

In your article "To Tell the Truth" [see the March 2000 issue of **EM**], you state that the Oktava MC012 is not available as a matched pair. I did some research and found that not only is a matched pair available, but it comes in a handsome cedar box. And can you beat the price? It's $629 for the pair, including several interchangeable capsules. I got this information on the Web from a place called the Sound Room (www.oktava.com).

Thank you for an otherwise extremely well written and informative article. It was very helpful.

Don Devine
Los Angeles, CA

MANLY WORDS

Bjorn Dittmer-Roche [see "Letters: A Little Respect" in the April

2000 issue of EM] may take offense at the "sexist" nature of some of the product ads in EM, but such is the nature of the music business. Pop music, which is often an expression of sexual themes, relies on technicians who passionately articulate their craft in similar terms. This male-dominated (and male-invented) field has produced its own colorful idiom.

Of the many talented female performers and writers I've met, none, not even the technically enlightened, ever got turned on by a cool piece of electronic gear. Conversely, I've never met a male engineer who hasn't at least once described a product using sexual language; for example, "That bass's got balls."

Is it surprising, then, that some manufacturers appeal to this idiom when advertising—and even naming—their products? Remember the Aphex Big Bottom derriere logo? Electric Mistress, anyone? What kind of woman is Lucille? The list goes on.

That there aren't more women in the audio field has nothing to do with how they are or are not "encouraged at a young age to fiddle with computers [and] 4-track recorders." Little girls, unlike little boys, naturally prefer toy dolls to toy trucks. Would feminists deny audio craftsmen their parlance in retribution for nature's disparity? Does the field really suffer from a lack of sexual diversity? I hope EM preserves its freedom in advertising, at least until as many women object to the naked-body ads in *Cosmo*.

Les Barker
Atlantic City, NJ

GRAND THEFT AUDIO

I've read many articles on sampling legality, and I'm pretty clear about using another artist's musical "bites." Legal permission, royalties, and credits for original performers are now commonplace for hip-hop CDs. But what about sampling dialogue from movies for use (often out of context) in my music?

Of the hundreds of recordings I own that use audio clips from movies (mostly industrial/electronica), not a single one lists the movie production companies or distributors. I'm talking about major-label artists such as Ministry and Skinny Puppy. They use samples from dozens of movies (from obscure,

B-grade horror films to big-studio classics), but they don't list performance rights in the liner notes. On the other hand, Information Society's debut album suffered a six-month delay while their label struggled to get the legal rights to use *Star Trek* dialogue in their music. (This may be a different scenario, though, because the samples that they used were from television episodes, not movies.)

I have some choice film tidbits "blended" into my CD, but I'm terrified of selling it on the Web and getting slapped with a lawsuit. How did other artists get away with it? Should I disguise the samples with filters, lower the pitch, bury them in the mix, and so forth? Or is sampling dialogue from films considered parody and fair use (in a fairly loose interpretation, of course)? Your insight is greatly appreciated.

Michael Stephens
Portland, OR

Michael—The movie or television production company is the copyright holder that owns all rights to the original work, including the exclusive rights to make copies or derivative works. When you sample an audio piece from a production, you're arguably infringing upon these rights. Additionally, the production company purchased the rights from the actors who performed on the original work, so any other use of the work for an unintended purpose (such as using a voice on a record when the performer only thought he or she was making a TV show) may open you and the original production company to potential liability from that actor.

Even taking into consideration the budget constraints of independents like yourself, I advise you to "clear" your sample by asking for permission. If you do go the route of "burying" the sample or changing it dramatically, an element of risk will still exist.—Michael A. Aczon

MAD SCIENTIST

Thanks for the great article on rack-mounting computers [see "DIY: Rack-Mounting Your PC" in the March 2000 issue of EM]. Another option is to use your existing case, provided it fits in your rack. Here's how I did it.

First I bought a piece of aluminum for the face. I then calculated how much to mill for the CD-R, CD-ROM, floppy, and removable drives. Then I

machined the back of the plate so that I could screw it to the computer case.

I'm thinking of turning this experiment into a full-time operation, building complete systems that include a keyboard, monitor, and mouse as well as the rack-mounted CPU. I'm even planning new designs that feature anodized covers.

Ivan Gomez
via e-mail

THE DEFENSE RESTS

It's very apparent from the "Letters" column in your April 2000 issue that you can't please everyone all the time. The petty criticisms of your magazine were quite irritating, to say the least. Every issue is well worth the money I put forth for it. I don't find every article useful for every situation, but your magazine has a wonderful way of spanning the amateur-to-professional spectrum and giving everyone something that can improve their recordings. For that I thank you.

On a side note, I think the January 2000 issue's black cover is fine [see "Letters: Cover Story" in the April 2000 issue of EM]. Black doesn't have to be a color of doom, gloom, and despair. It is rather the "fullness" of all colors mixed together. Philosophically speaking, it could be interpreted as the mixing of musical styles, cultures, and technology.

Some people throw the baby out with the bath water. My approach is to use what's beneficial, disregard what doesn't apply to me, and always remember that it's all about the vibe! Peace.

Mike
Wharton, TX

ERROR LOG

April 2000, "Out of the Bottle," p. 79: The third sentence should be "His past credits include Phil Collins's "Sussudio," Chaka Khan's "I Feel for You," and Steve Winwood's "Higher Love." David Frank's work on these songs included arranging and playing, not production.

Cool Edit Pro's got more effects than you can shake a stick at.*
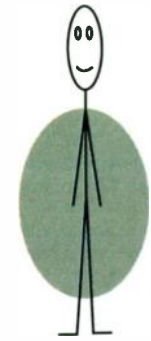
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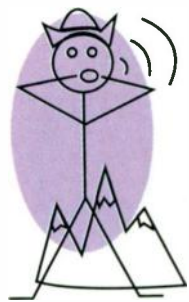
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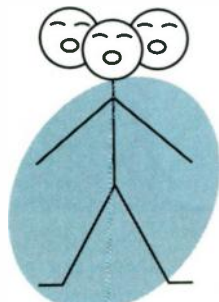
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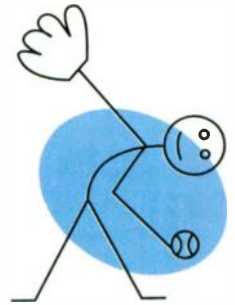
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Envelope Follower

Parametric EQ
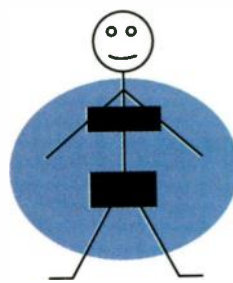
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*No stick-figures were hurt in the making of this advertisement.

WHAT'S NEW

By Barry Cleveland and Marty Cutler

▶ CAD C84

The sleek new CAD C84 condenser microphone ($199) is designed for applications for which high-frequency definition is particularly important. These include miking acoustic guitars and other stringed instruments, hi-hats, drum overheads, and choirs.

The transformerless C84 uses an onboard FET preamp and is externally biased, resulting in very high gain and, according to the manufacturer, low noise and distortion. It has a fixed-cardioid polar pattern and a wide frequency response.

The C84 microphone sports a gray finish with black accents and weighs 5 ounces. Frequency response is rated at 40 Hz to 16 kHz, with a dynamic range of 104 dB. The maximum SPL is 132 dB (1% THD, open circuit). Sensitivity is rated at 12 mV/Pa=38 dBV (0 dB=1v/Pa). CAD Professional Microphones; tel. (800) 762-9266; fax (440) 593-5395; e-mail jillo@cadmics.com; Web www.cadmics.com.

▼ AURALEX ACOUSTICS MAX-WALL

The Auralex Mobile Absorptive eXpandable (MAX) acoustic-treatment system is literally "off the wall." The MAX-Wall system ($349 per box) is modular and mobile, consisting of 2-by-4-by-4.375-inch interlocking panels that attach to tripod stands in various configurations. The system was designed for personal studios and other facilities where permanently attached acoustical treatment may not be desirable.

MAX-Wall units can be placed around the perimeter of a mixing area, effectively creating an acoustically treated "space" for near-field monitoring. They can also be used to create pseudo isolation booths for recording vocals or individual instruments; an optional Plexiglas window unit ($99) is available for that purpose.

The basic package contains four panels and two stands. As an added bonus, the stands come with boom arms, so you can use them as mic stands when they aren't supporting panels. Further options should be available in fall of this year, including stand-height extenders and a carrying case.

In addition, the company is shipping a stand-mounted version of the LENRD bass trap ($319 per box of four). These

traps are placed in the corners of your room to control bass frequencies, and Auralex recommends them as a complement to the MAX-Wall system. Auralex Acoustics; tel. (800) 95-WEDGE or (317) 842-2600; fax (317) 842-2760; e-mail auralex@auralex.com; Web www.auralex.com.

▼ HHB RADIUS 3

The 3-half-rackspace Radius 3 ($429) is a stereo tube compressor designed for those who are inexperienced with compression settings and would prefer to rely on customizable factory presets. The Radius 3 works in linked stereo mode but can be configured as a mono compressor by routing signals into the left input only.

The Radius 3's 15 presets are optimized for different types of vocals, guitars, drums, bass, and keyboards, along with overall stereo mix settings. Controls for input gain, output gain, and makeup gain remain active while using the presets, and a Manual Mode setting allows adjustment of threshold, ratio, attack (fast or slow), and release (fast or slow) settings. A single VU meter monitors either gain reduction or output level.

The Radius 3 has balanced line inputs and outputs on ¼-inch TRS jacks, and operating levels are switchable between +4 dBu and −10 dBm, making the unit suitable for use in both professional and semiprofessional applications. HHB Communications USA; tel. (310) 319-1111; fax (310) 319-1311; e-mail sales@hhbusa.com; Web www.hhb.co.uk.

▶ HAFLER P4000

The 3-rackspace P4000 Transnova DIAMOND (Dynamic Invariant Amplification Optimized Nodal Drive) stereo power amplifier ($999) is optimized for studio applications and features Hafler's respected Transnova circuitry. The Transconductance driver stage combines Class A linearity with the current headroom of Class B circuitry.

The P4000 is convection cooled and does not have a fan. It delivers 200 watts per channel into 8 ohms, 275 watts per channel into 4 ohms, or 550 watts bridged mono into 8 ohms. LEDs indicate power on, signal present, clip, thermal, and protect. The gain controls range from –29 dB to +29 dB. The amplifier accepts both balanced inputs via XLR and ¼-inch TRS connectors and uses lateral MOSFET output devices and gold-plated 5-way binding posts.

The 34-pound P4000 boasts a frequency response of 20 Hz to 20 kHz, <0.2% THD, and a 100 dB S/N ratio (A weighted). Hafler; tel. (888) HAFLER1 or (480) 967-3565; fax (480) 894-1528; e-mail sales@hafler .com; Web www.hafler.com.

▶ ROLAND HPD-15

Roland has ported elements of its V-drum technology to the HPD-15 Handsonic Hand Percussion Pad ($1,295). A 10-inch pad divided into 15 parts is used to trigger up to 15 simultaneous sounds, and striking on different areas of the pad creates variations in timbre. A built-in pressure sensor can be used for expressive real-time control of parameters, including pitch modulation. With Roland's D-Beam, the player can trigger sounds or change instrument tuning in real time. Additional controllers include two ribbon controllers and three knobs.

The new controller's built-in sequencer has a number of preset patterns covering a wide range of musical styles. A user-memory area allows recording and storing of real-time performances. MIDI In and Out jacks access external devices and sequencers. You can play the unit from external sources with the trigger input; an expression pedal input can be used for continuous variation of the hi-hat sounds.

The HPD-15 features 300 percussion sounds from all parts of the world. Additional sounds include acoustic and electronic drums, sound effects, and a number of backing instruments (pianos, guitars, basses, and more). There are 80 user patches for storage of customized sounds. Onboard effects, including a dedicated reverb and a multi-effects processor, add extra polish to the sounds. Maximum polyphony is 64 voices. Roland Corporation U.S.; tel. (323) 890-3700; fax (323) 890-3701; Web www.rolandus.com.

▼ OBERHEIM OB-12

Oberheim has kicked off a new product line with its first analog modeling synthesizer, the OB-12 ($1,999). Encased in an attractive bright-blue chassis, the 12-voice synthesizer features 24 sliders, 22 knobs, and 33 buttons for real-time control of timbre. Five buttons and four knobs can be used to control both the levels and parameters of the built-in effects.

The OB-12's voice architecture offers two oscillators that can generate sawtooth, square, or triangle waves, which can be blended with the WaveMix feature. You also get one noise generator and one ring modulator per voice. Pulse width is variable for the square wave on oscillator 2, and the sawtooth wave can be detuned. The triangle waveshape can be modulated to create new waveforms. LFOs use sawtooth, triangle, square, and sample-and-hold waveforms.

You can configure the two 24 dB-per-octave filters in serial, parallel, and split modes. Envelopes go beyond standard ADSR control, adding a Delay Time stage and a second Decay Time stage. Onboard effects include overdrive, chorus, delay, and reverb.

All edited parameters, including envelopes, are graphically displayed on a 240-by-64-pixel LCD. Parameter changes can also be captured in the OB-12's Motion and Phrase Recorder and can slave to MIDI Clock. The unit offers a stereo S/PDIF digital output, in addition to the left/right analog outputs. Armadillo Enterprises (distributor); tel. (727) 519-9669; fax (727) 519-9703; e-mail armadillo@ packet.net; Web www .oberheim.com.

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

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- 32-bit audio signal processing
- Reads GigaSampler, .WAV, and Akai™ Libraries
- Includes GigaPiano™ Sample Library CD-ROM

USB CONTROL SURFACES ▲ ▲ ▲ ▲

▲ TASCAM US-428

Designed to present a familiar face to Portastudio users, Tascam's US-428 24-bit digital audio workstation controller ($599) combines a control surface, MIDI interface, and computer audio interface for Mac and Windows computers equipped with USB connections.

The unit provides mixing and transport controls for digital audio workstations and sequencers. The audio interface provides four 24-bit analog input channels: channels A and B have mic-level balanced XLR jacks and line-level balanced ¼-inch inputs.

Channels C and D offer switchable analog or digital I/O. The analog inputs are on two unbalanced ¼-inch jacks with a guitar-input switch.

You also get stereo S/PDIF coax digital input and can record up to four channels simultaneously, with 24- or 16-bit resolution. Sampling rates of 48 kHz and 44.1 kHz are supported. Inputs C and D are switchable between analog and digital input.

Analog outputs are on unbalanced RCA line-out jacks; digital outputs are coaxial S/PDIF. Two MIDI inputs and outputs provide 32 channels of MIDI control.

Eight hardware faders allow control of multiple MIDI channels and multiple groups or banks of software faders. EQ controls support up to four bands of parametric EQ. You also get controls for aux sends, pan, and the recorder transport. A set of programmable keys and a data wheel allow you to control your choice of software functions. Tascam/

TEAC; tel. (323) 726-0303; fax (323) 727-7635; faxback (800) 827-2268; Web www.tascam.com.

▼ EDIROL U-8

Edirol is now shipping its U-8 USB Digital Studio ($795), a 24-bit, 44.1 kHz hard disk recording system for Windows. The U-8 is available bundled with either Steinberg *Cubasis* or Cakewalk *Home Studio* 9 software.

The U-8 integrates analog and digital I/O with built-in 24-bit DSP and a control surface. The control surface features assignable sliders, faders, pan pots, and transport controls that offer direct control of the software's mixing functions. An interactive setup procedure guides you through the recording setup process, from choice of input to adjustment of recording levels.

The U-8 comes with more than 130 onboard, 24-bit DSP effects, including reverb, amp simulation, compression EQ, distortion, and noise suppression. You can adjust every individual parameter through the U-8's three assignable effects knobs.

The unit provides an unbalanced XLR mic input, unbalanced ¼-inch guitar and line inputs, two RCA inputs, and two RCA outputs. Optical S/PDIF digital I/O is provided, and you get one MIDI In and one MIDI Out jack. You'll need at least a Pentium II/300 MHz with 64 MB of RAM, Windows 98 or 98 SE, and a USB port. Edirol North America; tel. (800) 380-2580 or (360) 332-4211; fax (360) 332-4405; e-mail edirol@edirol.com; Web www.edirol.com.
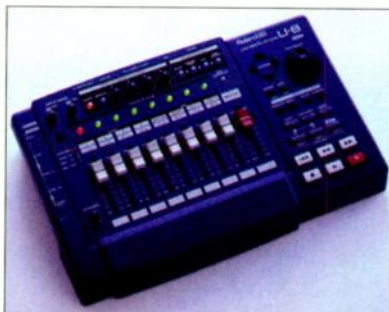
▼ RADIKAL TECHNOLOGIES SAC-2K

The Radikal Technologies SAC-2K ($1,795) offers a control surface for the computer-based digital audio workstation that goes beyond the standard mixing-surface metaphor. The touch of a button turns the SAC-2K into a remote synthesizer-programming surface, either as a stand-alone editor or in conjunction with editor/librarian software. Real-time synth-parameter changes can be recorded into a sequencer using the unit's programming interface.

The SAC-2K accomplishes its tasks with rotary encoders whose assigned parameters are displayed on a large, 2-line LCD. A dedicated set of LED-illuminated switches controls track arming and solo or mute functions. Nine 100-millimeter motorized faders can be assigned and grouped in any combination, and you can control them individually or in groups (up to eight at once). A central array of buttons controls effects, EQ settings, sends, returns, busing assignments, panning, and selection of groups. The Musical Instrument button converts the SAC-2K into a System Exclusive editor for synthesizers.
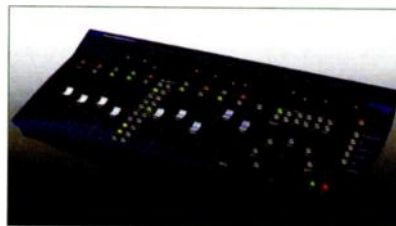
The transport controls (forward, reverse stop, play, and record) are centrally located. You can store and recall markers, select ranges for editing, enter numerical values, and control typical computer functions such as undo and file saving. Radikal Technologies; tel. (201) 836-5116; fax (201) 836-0661; e-mail jsk1@gramercy.ios.com; Web www.radikaltechnologies.com.

KEY CHANGES

BitHeadz has released a version of *Steve Reid's Percussion Kit* ($199) for the company's *Unity DS-1* software sampler and *Voodoo* software drum machine. The collection provides more than 200 MB of percussion samples . . . **Sonic Foundry**'s *Acid Music* 2.0 ($99.95) now features CD burning capability and supports MP3, RealMedia G2, and Windows Media formats. New effects include reverb, chorus, delay, and EQ. *Acid Pro* 2.0 ($399.95) now comes with the Sonic Foundry *XFX1* DirectX plug-in, and is bundled with *Sound Forge XP 4.5* . . . **Creative Technology** has released a Mac version of the Sound Blaster Live audio card ($199.99) . . . **Cakewalk**'s *Pro Suite* ($829) bundles Cakewalk *Pro Audio 9* with **NemeSys**'s *GigaSampler LE* software sampler and Cakewalk's *Audio FX 1 Dynamics Processor*, *Audio FX 2 Vintage Analog Simulator*, and *Audio FX 3 Sound-Stage Designer* DirectX plug-ins. The included 2-CD *Musician's Toolbox* provides more than 1 GB of audio and MIDI loops, DNA groove templates, banks of Sound-Fonts, digital video libraries, and more . . . **Spirit by Soundcraft** has released version 1.5 software for its Digital 328 mixer. Among its new features, the upgrade lets you send MIDI Control Change messages and has 64 memory slots for naming and storing MIDI control setups. It can be downloaded free from the company's Web site (www.spirit-by-soundcraft .co.uk) . . . **Korg USA** has released new operating systems for its I-series synths. OS 2.13 for the I30 operating system can sort and search for Standard MIDI Files by name, edit the ROM bank, provide real-time slider control of the mixer during playback, and more. OS 3.0 for the IS40 and IS50 lets you preview arrangements and programs before selection and offers drum remapping options, compatibility with Arabic scales, and more. OS 3.0 for the I40M includes all of these features and adds automatic vocal- and guitar-input setup using an A1 keyboard set.

▶ FOSTEX D1824 AND D824

Fostex has entered the arena of 24-bit, 96 kHz multitrack hard disk recording with two new models. The D1624 ($3,095 with hard drive, $2,795 without) is a 16-track modular hard disk recorder, and the D824 ($2,095 with hard drive, $1,795 without) records 8 tracks. Both units can record uncompressed 24-bit or 16-bit audio at 96, 48, or 44.1 kHz. The recorders sport an internal caddy that can hold a standard EIDE hard drive.

You can load songs in WAV file format, and each song can use up to 16 virtual tracks (with the D1624) or 8 tracks (with the D824). Audio editing is nondestructive, and material can be copied and pasted from one song to another. The editor features as many levels of undo as hard disk space permits, and you get 6 edit points and 99 location points. Multiple time signatures and tempos are supported for each song.
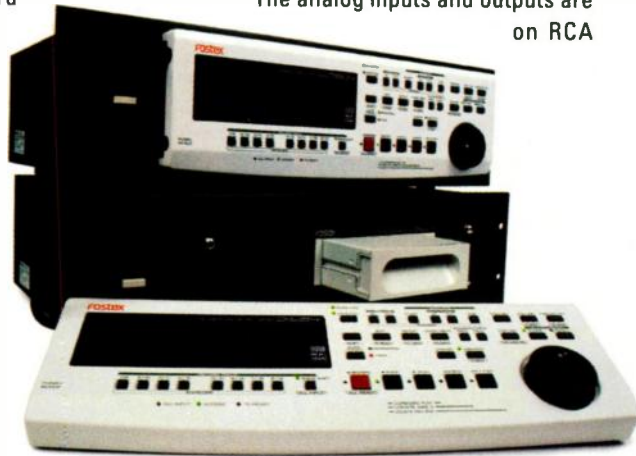
Both units send and can sync to MIDI Clock. An RS-422 remote-control connector is included, as is SCSI. Digital I/O is via S/PDIF and ADAT optical connections. The analog inputs and outputs are on RCA connectors. Time code, AES/EBU, and balanced I/O are available as options. Fostex of America; tel. (562) 921-1112; fax (562) 802-1964; e-mail info@fostex.com; Web www.fostex.com.

▼ STEINBERG PPG WAVE 2.V

The PPG Wave 2 was the first digital synthesizer that could create extremely animated timbres by sweeping through a series of waveforms (which PPG called *wavetables*). A physically modeled software version of this vintage synthesizer is now available with Steinberg's release of *PPG Wave 2.V* ($199).

A software synthesizer integrated into the VST environment, *PPG Wave 2.V* can be processed through the VST mixer, so that effects, EQ, and other VST plug-ins can be applied to the synth's outputs, and you can save all settings within *Cubase* song files. MIDI tracks are internally routed to the synthesizer. Up to eight synths can be open simultaneously, and each can play up to eight multitimbral parts. Polyphony is dependent on your computer's processor speed.

PPG Wave 2.V features 32 wavetables (with up to 64 waveforms per wavetable), two oscillators per voice, and a lowpass filter with emphasis (resonance). Envelope generators, as well as filter cutoff and emphasis parameters, can be edited with a graphical interface.

The cross-platform program requires a VST 2.0–compatible host program. PC users need a Pentium II/266 MHz or faster processor (400 MHz recommended), 64 MB of free RAM, and Windows 95 or 98.

Macintosh users must have at least a 604e/250 MHz processor (G3 recommended), 64 MB of free RAM, and Mac OS 8.0 or higher. Steinberg North America; tel. (818) 678-5100; fax (818) 678-5199; e-mail info@steinberg-na.com; Web www.us .steinberg.net; www.cubase.net.

SOUND ADVICE ▲ ▲ ▲ ▲

ALTERNATE MODE

Designed to work with E-mu and EOS samplers and drum controllers, *The Arthurian World of Percussion* ($199) from Alternate Mode is a collection of more than 125 percussion instruments from around the world. All of the sounds were taken from the collection of Drum Circle percussionist Arthur Hull.

To preserve the drums' nuances, the sounds have been re-created using multiple samples that were programmed with Velocity switches and Velocity crossfades. Each instrument in the 640 MB collection is automatically loaded into the same MIDI keymap layout.

Alternate Mode; tel. (413) 594-5190; fax (413) 592-7987; e-mail kat1993@aol.com; Web www.alternatemode.com.

▲ STEINBERG

Steinberg has released a new set of drum sounds to support its *LM-4* virtual drum machine. Compiled from sounds by BitBeats, the XXL set ($49) contains natural dry kits recorded in stereo, atmospheric-sounding sets recorded with room ambience, pure electronic sounds, and assorted percussive effects. The 36 kits are divided into Dry Kits, Ambient Drums, Power Drums, Synth Drums, FX Drums, and Percussion. Steinberg North America;

tel. (818) 678-5100; fax (818) 678-5199; e-mail info@steinberg-na.com; Web www.us.steinberg.net.

▼ GEFEN

Gefen's Renaissance collection ($895) offers seven CDs with ambiences, sounds from nature, special environments, music tools, and more. Gefen Systems; tel. (800) 545-6900 or (818) 884-6294; fax (818) 884-3108; e-mail gsinfo@gefen.com; Web www.gefen.com.

▶ MINIMUSIC BEATPAD

BeatPad ($29.95) from Mini-Music is a pattern-based sequencer for Palm computers. You can enter and edit notes (even during playback) using a simple piano roll–style or drum-pattern editor. A drum pattern can contain up to 11 instruments, and you can edit pattern dynamics in the accent track. *BeatPad* can play back a single melodic pattern and a drum pattern, and you can change or swap a pattern with another pattern in real time. Accents, pitch, and durations can be entered using the Palm stylus.

BeatPad plays back through the Palm's built-in speaker (albeit without drum sounds) or any serial-port MIDI interface and synthesizer using a Palm serial computer connector. Using Palm's Hot-Sync technology, you can sync files created in *BeatPad* with Standard MIDI Files on your desktop computer so that the SMFs can be imported and exported. MiniMusic; tel. (415) 831-1514; e-mail support@5thwall.com; Web www.5thwall.com/minimusic.

▶ JOHNSON J-STATION

Johnson Amplification's J-Station is a compact tabletop effects processor tailored for guitarists. The J-Station offers digital models of several classic guitar and bass amps and an acoustic guitar emulation model. There are separate sections for amp models and effects, and six dedicated knobs for gain, treble, mid, bass, level, and master volume.

The J-Station provides a host of programmable effects including compression, reverb, chorus, wah-wah, and pitch-shifting. You can quickly adjust effects to your taste with the three dedicated knobs, and effects can be stored in any of the 30 user-memory locations. The J-Station's MIDI input and output enable changing patches via MIDI Program Change commands. Users who want to dig deeper into customizing the J-Station can use the supplied editor/librarian software for Windows. You can also share patches with other J-Station users through System Exclusive. Audio outputs are on ¼-inch left/right unbalanced jacks, and a coaxial S/PDIF output allows recording directly into digital recording devices. The J-Station can also serve as a preamp and effects processor when used with an amplifier in a guitar or bass rig. Johnson Amplification; tel. (801) 566-8800; fax (801) 566-7005; Web www.johnsonamp.com. ⊕

Illustration by Dennis Irwin

Double Your Pleasure

Whether you record live or in the studio, stereo recording can heighten the realism of your productions. Why is that? Simply put, humans have two ears, so stereo-sound reproduction sounds more natural to us than mono does. In fact, most stereo-miking setups grew out of attempts to approximate the positioning and functioning of our ears.

Another reason that stereo recordings usually sound more natural than mono recordings or stereo mixes made from multitracked mono sources is that stereo recording is often done ambiently—that is, with the mics placed at a distance from the sound source rather than directly next to it. Obviously, we don't usually listen to instruments with our ears right up next to them. Close-miking can result not only in an unnatural-sounding magnification of detail, but also in tonal imbalances such as overly bright tones, sibilance, boominess, and the like.

In short, ambient stereo-miking tends to capture the natural characteristics of an instrument better than close mono-miking. Done well, it conveys localization (left-right placement in the stereo field) and, in comparison to mono, superior depth of field (the sense of distance from the instruments) and a fuller, more accurate "picture" of the recording space's acoustic properties (including reverberation, diffusion, and other elements).

By Elizabeth Papapetrou

Double Your Pleasure
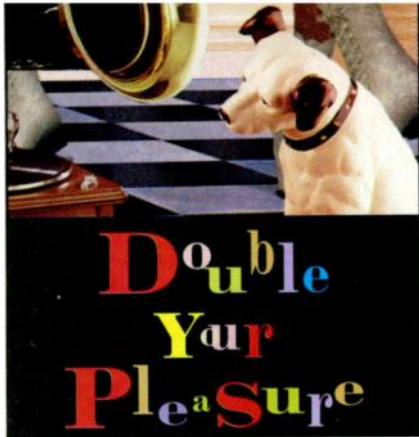
Stereo-miking techniques can be broken down into three types, defined by how the microphones are positioned: coincident, near-coincident, and spaced. There are also a number of dedicated stereo microphones available that offer two or more diaphragms in the same mic body (for a variety of coincident-miking techniques), as well as stereo-miking systems such as the Crown SASS-P (Stereo Ambient Sampling System). This article, however, focuses on techniques that use individual microphones (usually two, but in some cases, three or more).

COINCIDENT

As the name suggests, a coincident pair is two microphones positioned so that their capsules are as close to each other as possible. This closeness, or coincidence, ensures that the left and right signals have no phase differences and thus maintain their frequency response when summed to mono.

In coincident pairs, the stereo spread is created by amplitude alone. Directional microphones are most sensitive to sound coming from directly in front of the capsules (on axis) and are less sensitive to off-axis sound. In other words, the mic "hears" lots of sound

FIG. 1: XY pairs of mics are typically angled 90 to 135 degrees apart. Because no time differences occur between pickup by the two capsules, amplitude alone accounts for spatial positioning in the stereo field.

coming from the front and much less from the sides. Therefore, sound sources positioned in the center of the angle between the two mics will appear in the center of the stereo field, whereas sources off to one side or the other will appear more on-axis to one microphone and thus will be reproduced louder on that side.

One of the simplest and most popular coincident-pair techniques is the XY pair. For balanced stereo reproduction, the two mics should be the same model and preferably matched (meaning that the manufacturer has taken extra care to ensure identical functioning). Also, it's important that the mics have directional polar patterns, whether cardioid, hypercardioid, or supercardioid.

The angle between the two capsules in an XY pair is usually set between 90 and 135 degrees (see **Fig. 1**). The wider the angle, the broader the stereo spread. However, compared with other stereo-miking techniques, even miking with a wide-angle XY pair will produce a rather narrow, focused stereo image generally lacking in depth and "airiness." In addition, there may be sound coloration due to the fact that off-axis sound constitutes a significant portion of pickup. (This can be a problem with some microphones and should be taken into account.)

When you're recording multiple instruments, you can widen the stereo spread by arranging the players in a semicircle around the XY pair. XY pairs also work well for close-miking instruments such as acoustic guitar and piano.

Naturally, the angle of the mics also affects the capture of reverberation and other ambient elements. An XY pair angled 90 degrees apart, for instance, places most of the room reverberation in the center; wider angles spread the reverberation across the stereo field.

BLUMLEIN PAIR

A very effective variation on the coincident-
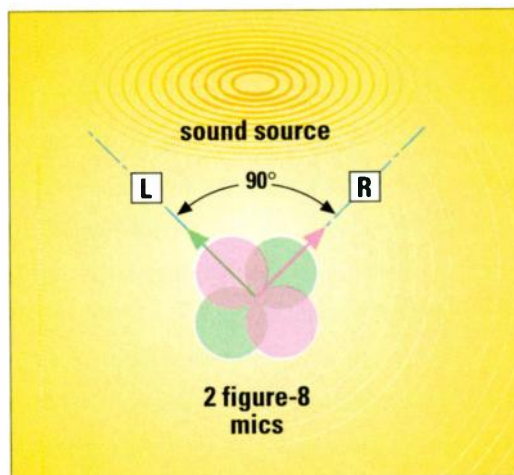
FIG. 2: Two figure-8-pattern mics angled 90 degrees apart form a Blumlein array. This technique is especially effective when a broad pickup of room sound is desired.

cardioid pair is the Blumlein coincident pair, named after Alan Dower Blumlein, the pioneering electrical engineer and inventor who developed the technique. This setup requires two bidirectional mics (that is, mics with figure-8 polar patterns) whose capsules are angled 90 degrees apart. As with coincident-cardioid pairs, the sound source is centered between the two front-facing capsules (see **Fig. 2**). Essentially, the Blumlein array creates a four-sided polar pattern that gets summed to two channels. The left channel is made up of front-left and rear-right signals, and the right channel consists of front-right and rear-left signals.

To my ears, the stereo imaging captured by a Blumlein pair is hard to beat—the clarity and detail of stereo localization are simply nonpareil. (For information about stereo localization, see the sidebar "At Play in the Stereo Fields.") Another distinguishing characteristic is the uniform spread of room reverberation across the stereo field. Nonetheless, the use of Blumlein arrays does have some minor drawbacks.

One common problem is excessive room sound. To compensate, engineers often position the mics closer to the sound source, thereby increasing the ratio between direct and reverberant sound. This compensation, however, can overemphasize pickup from the center of the source, which in turn can make it harder to pinpoint the locations of sound sources at the far ends of the stereo spread (that is, at "the edges" of the 90-degree arc defined by the front-facing polar patterns). In other

Double Your Pleasure

words, as you move the mics closer to the source, the stereo spread starts getting too wide.

Blumlein pairs also present the risk of unwanted phase anomalies, particularly when recording a sound source (for instance, a large ensemble) that stretches beyond 45 degrees from center. In such situations, phase anomalies are more likely to occur because the sound is being picked up by one mic's front diaphragm and the other mic's rear diaphragm, which are out of phase with one another. Again, the solution is to make sure that the sound source falls within the 90-degree arc defined by the front-facing microphone diaphragms.

For these reasons—especially if naturalness of sound is a primary concern (as it is when you're recording a string quartet, for example)—I wouldn't recommend using a Blumlein array in a highly reflective room or in a narrow room with a lot of sidewall reflection. On the other hand, if you want to capture lots of room sound (when recording a guitar amp, say), a Blumlein pair may be just what the doctor ordered.

MIDDLE-SIDE PAIR

A middle-side (M-S) stereo pair employs one "mid" microphone (typically a directional mic with a cardioid pattern) that faces the sound source and one bidirectional "side" mic. The two mics are positioned so that the mid mic's primary axis is aligned with the side mic's null axis (the dead area on either side of the figure-8 pattern). This arrangement (see **Fig. 3**) captures a very stable center image and has the additional advantage of being thoroughly mono compatible.

To create a complete stereo image from M-S signals, you must "decode" them with a *sum-and-difference matrix*. The matrix, in a typical M-S setup, produces the left and right channels from the middle and side signals using the following equations (assuming the side mic's positive lobe is oriented to the left):

left channel = middle + side
right channel = middle − side

One common way to decode M-S signals is to use three channels on a mixing board: the mid mic's signal is panned dead center in one channel and the figure-8 mic's signal is split (using a Y-cable) to feed two channels, one panned hard left and the other hard right (see **Fig. 4**). To complete the process, the polarity of one of the "side" channels is then reversed, either with a polarity-reversal switch on the console channel or with a special "polarity reversal" cable (one whose "hot" and "cold" wires have been reversed).

One potential problem with this form of sum-and-difference matrix, howev-
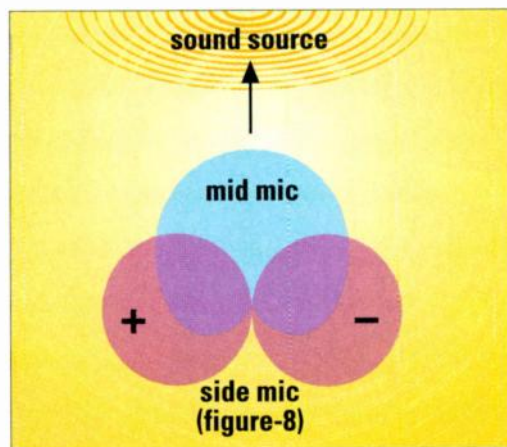
FIG. 3: This diagram shows the combined polar patterns of a middle-side array.

er, is that the side microphone is driving twice its normal load at only half its normal termination impedance—an imbalance that can result in distortion and a higher noise floor. For that reason, engineers who specialize in M-S recordings often employ gear with dedicated sum-and-difference-matrix capabilities. Units with active components are favored, as these provide a well-balanced matrix and, typically, a control system to vary the M-to-S ratio without damaging the signal-to-noise ratio, phase response, or frequency response. Many companies—for example, Calrec, E.M.T., Soundcraft, Telefunken, and Yamaha—have manufactured active M-S console modules. Currently available stand-alone models include Grace Designs' Lunatec V2 and Audio Engineering Associates' MS38 mkII, an outboard, active, line-level matrix control unit that provides a single M-to-S ratio control. Also on the market are stereo mics designed to record M-S and automatically decode the signals; for example, the Shure VP 88.

Part of the appeal of M-S recording is that you can manipulate the stereo image without having to rearrange the mics—even after the recording is completed. For example, you can tighten the stereo image and make the source seem closer by increasing the level of the mid-mic signal in relation to the side-mic signals. Conversely, increasing the level of the side-mic signals makes the source seem more distant.

The stereo image and overall sound can also be altered through panning: as the side-mic signals are panned closer to center, the stereo image gets narrower and the mid-mic signal gets

A WALK ON THE MID-SIDE

Middle-side (M-S) stereo recording was developed for stereo broadcasts in the 1950s by Danish radio engineer Holger Lauridsen. A popular technique in Europe, it was relatively unknown in the United States until the advent of stereo TV broadcasts.

Engineers Wes Dooley and Ron Streicher popularized the technique in the United States with their widely read technical papers and their use of M-S in the 1980 Emmy Award–winning broadcast of Beethoven's Ninth Symphony (performed by the Los Angeles Philharmonic and the Los Angeles Master Chorale and broadcast by KCET, channel 28 in Los Angeles). M-S recording is now used by many diverse organizations, including National Public Radio, Louisiana State University, and cinema-sound facilities such as SounDeluxe and Lucasfilm's Sprocket Systems.

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louder, making the source seem closer. Panning the left and right side-mic channels to dead center (that is, to mono) effectively removes the side-mic signals and increases the mid-mic signal by 6 dB. The equation is $(M + S) + (M - S) = 2M$.

If a bidirectional mic is not available, you can construct a passable M-S setup by substituting a matched pair of cardioid microphones placed back-to-back (for the two side signals) with the polarity reversed on one of the mics. However, this method does not work as well as a true M-S setup. (For information about the origins of M-S recording, see the sidebar "A Walk on the Mid-Side.")

NEAR-COINCIDENT

A near-coincident stereo array uses two directional microphones angled away from one another, with their grilles spaced a few inches apart in the horizontal plane (see **Fig. 5**). Again, the angle between the mics causes an intensity or level difference that creates the sense of the sound source's position in the stereo field. In addition, the space between the diaphragms introduces a time or phase difference to the stereo information, adding width and depth to the stereo spread and airiness to the recording. With near-coincident pairs, then, both amplitude and time difference contribute to stereo localization.

Results of near-coincident miking can vary considerably, depending on the frequency and polar-pattern (usually cardioid) characteristics of the particular microphones as well as on other factors. In general, watch out for too great an angle or space between the mics, as this will cause exaggerated separation. Similarly, too small an angle or space may result in an overly narrow stereo spread.

ORTF

One of the most widely used and reliable near-coincident arrays is ORTF, so called because of its early use by the

AT PLAY IN THE STEREO FIELDS

Stereo localization—defined here as the placement of sound sources in the recorded stereo field relative to their true positions during performance—varies in accuracy from one stereo-recording technique to the next. To compare the localization of the different techniques, stereo-miking expert Bruce Bartlett carried out a listening test using ten different stereo arrays (see *On-Location Recording Techniques* by Bruce Bartlett and Jenny Bartlett, Focal Press, 1999).

With each microphone array, Bartlett recorded speech sources at 0 degrees (center), 22.5 degrees (both left and right of center), and 45 degrees (also left and right). He did the test recordings twice: once in an anechoic chamber and again in a reverberant gymnasium. Afterward he played the results of each miking technique and asked his listeners to locate the sound sources' positions in the stereo field.

This table shows the averaged test results (the chamber and the gym recordings, by the way, were very similar). The upper portion shows the locations of the sound sources (labeled A through E). The bottom portion indicates where the listeners perceived them to be.

top view — 22.5° — 45° — center of mic array

perceived image locations		left speaker	half left	center	half right	right speaker
coincident	cardioids angled 90°		A	B C D	E	
	cardioids angled 135°		A	B C	D	E
	cardioids angled 180°	A	B	C	D	E
	figure-8s angled 90° (Blumlein)	A	B	C	D	E
near coincident	cardioids angled 90°, spaced 8"		A B	C	D	E
	cardioids angled 110°, spaced 6.69" (ORTF)		A	B C	D	E
spaced	omnis spaced 6.5" with baffle (OSS)	A		B C	D	E
	omnis spaced 3'	A	B	C	D	E
	omnis spaced 10'	A	B	C	D	E
	3 omnis spaced 5' (Decca Tree)	A	B	C	D	E

RENDERED FROM FIGURE 10-2 ON PAGE 115 OF *ON-LOCATION RECORDING TECHNIQUES*, BRUCE BARTLETT AND JENNY BARTLETT, FOCAL PRESS, 1999.

So you need a USB MIDI interface for your iMac, G3 or PC?

We've got a solution for you.

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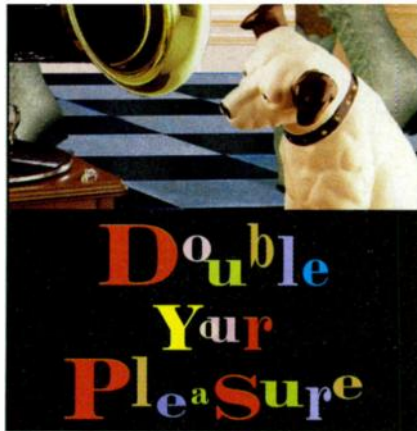
Office de Radiodiffusion Télévision Française. The ORTF technique specifies two cardioid mics with their capsules spaced 6.69 inches apart (the average distance between an adult's ears) and angled 110 degrees apart (see **Fig. 6**). Designed to mimic human hearing, ORTF provides a wide, accurate image and a good sense of depth. However, if the two signals are summed to mono, phase cancellations occur, which can cause comb-filtering effects that color the sound considerably. Yet it's the slight time delay between the left and right channels that contributes to the sense of stereo spread; if you don't need to reproduce the recording in mono, the ORTF technique will usually yield excellent results.

OSS

The OSS (Optimal Stereo Signal) system was created in the 1980s by Swiss
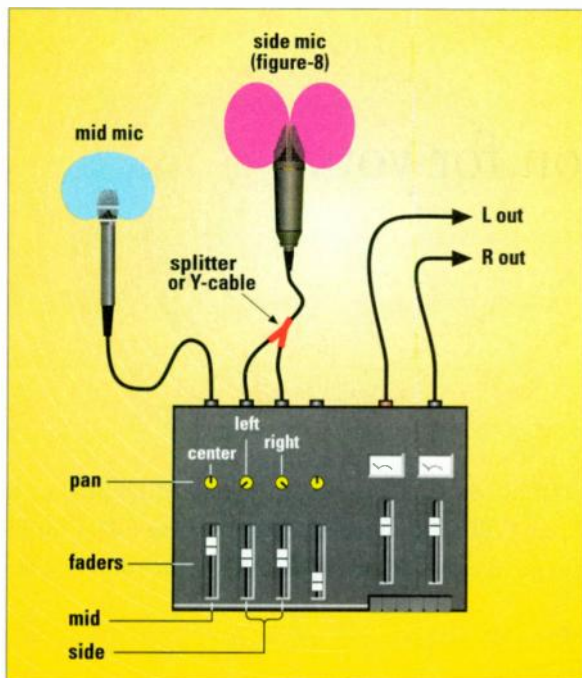
engineer Juerg Jecklin. The setup specifies a pair of omnidirectional microphones whose capsules are positioned 6.5 inches apart and separated by a sound-absorbing disk—called a *Jecklin disk*—measuring 11.81 inches in diameter. The microphone capsules are aligned with the center of the disk (see **Fig. 7**). Jecklin's original disk, made from an LP record, was covered on both sides with lamb's fleece, which is an excellent sound absorber.

Like ORTF, OSS is designed to simulate the positioning and pickup characteristics of human ears. The Jecklin disk takes OSS's simulation a step beyond ORTF's, providing separation and interference similar to that which is created by the human head. In large part, it is the addition of the disk that warrants the adjective *optimal* in *Optimal Stereo Signal*. When we listen to a sound source directly, our brains determine its location by analyzing the combination of three elements: sound intensity (level), time delays (phase relationships), and subtle variations in the way that different frequency ranges are picked up.

Near-coincident pairs employ only those first two elements to create stereo localization. Jecklin's OSS system, however, integrates all three elements to produce very lifelike stereo imaging. Note that OSS also utilizes omnidirectional, rather than unidirectional, microphones, which act much like our ears, picking up frequencies below 200 Hz equally. But as the frequencies increase (that is, as the pitches get higher), the disk (like a human head) blocks more sound, and the mics (like human ears) increasingly operate more directionally.

The OSS technique captures a full-bodied stereo spread rich in
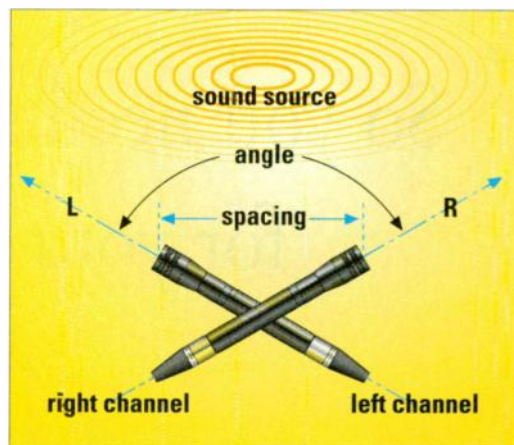
FIG. 5: In a near-coincident stereo array, the capsules of two directional microphones are spaced a few inches apart and angled outward. The perceived spatial position of the source is achieved through sound intensity as well as time (phase) differences.

detail. However, the same qualities that allow "optimal" stereo signals make it impossible to sum those signals to mono without introducing phase distortion.

Building your own Jecklin disk is fairly easy (see **Fig. 8** for a visual reference). Use wood, plastic, or a similarly stiff material (maybe a vinyl LP!) for the disk, which must be sturdy enough to serve as a mounting surface for the mic clips and mic stand. Cover both sides with at least 0.5 inches of acoustic foam. (I've also used several mouse pads glued together.) Mount the mic clips opposite each other near one edge of the disk so that the mic capsules are 6.5 inches apart and aligned with the center of the disk. Attach a mic-stand mount to the bottom of the disk, and you're ready to go.

BINAURAL RECORDING

Binaural recording is clearly the stereo-recording technique that best approximates human hearing. Like OSS, it employs interference between the mic capsules—but in this case it's caused by a replica of a human head rather than by a disk (see the sidebar "Background on Binaural"). Binaural recordings are made using two tiny omnidirectional mics placed in the dummy head's ears (or, for even greater realism, using special mics placed in the ears of a human subject). The two signals are kept separate throughout the signal path from the mics to the corresponding left and right drivers of the listener's headphones. Other than the binaural source

FIG. 4: An M-S array requires that you use a sum-and-difference matrix to convert the three signals to stereo. The operation can be done using three channels of a mixer, as shown. Note the Y-cable and the polarity reversal of one "side" signal.

Double Your Pleasure

and a pair of stereo headphones, no special equipment is required to convey the binaural experience to the listener.

Conventional binaural recording is generally incompatible with stereo loudspeaker reproduction and can sound strange. However, some modern binaural setups employ equalization and special design features to deliver signals that sound good on stereo speakers. Even with the best setups, though, not all of the directional material translates to speakers; the eerie "you are there" binaural effect is largely lost, due to the leakage of sound cues intended for one ear into the other ear.

You can therefore best experience binaural recordings with headphones, through which the full 360-degree sphere of pickup can be heard. Sounds are then discernible from any location—right, left, front, back, up, and down—and their distance from the mics can also be clearly made out. The sense of realism is uncanny: a good rainstorm recorded binaurally can

make you think that you're getting soaked.

Binaural recording works like this: the *pinna*, or outer ears, of the dummy head (or of the human subject wearing in-ear microphones) set up subtle interference patterns that give specific location to the sounds around the mics, thus positioning them accurately in space. Sounds coming from directly in front of the head, for example, bounce off the rear of the outer ear, sounds from below bounce off the top of the outer ear, and so forth. These ear-reflected sounds, in conjunction with sound that enters the ear canal directly, interrelate to create the interference patterns (known as Head Related Transfer Functions, or HRTFs) that the brain uses to determine sound location.

Binaural recordings are easily resolved to 5.1 and 7.1 (or any ".1," for that matter) surround sound and consequently are likely to become more prevalent in the audio-recording world. Binaural recording will also play a pivotal role in the development of virtual reality for computer games and other applications. Even for conventional stereo recording and playback, some engineers are of the opinion that modern binaural methods provide better-balanced and more-natural-sounding recordings than other stereo-miking techniques.

SPACED

A spaced, or AB, stereo pair (see **Fig. 9**) consists of two identical microphones placed several feet apart and aimed directly at a sound source. The mics can have (or be set to) any polar pattern, but obviously the pattern must be the same for each mic. In addition, to maintain consistency of pickup, the mics are usually positioned parallel to one another. Not surprisingly, the stereo spread increases with the distance between the two mics.

Stereo localization in spaced pairs, unlike that in coinci-

FIG. 7: In an OSS setup, the Jecklin disk acts as a barrier to create interference—much like a human head between two ears. And in another concession to human physiology, the mics are single-point omnis rather than directional mics.

dent and near-coincident arrays, is provided by time delay only. Sound from directly in front of the mics reaches both capsules simultaneously, so the recorded sound appears in the center of the stereo field. Off-center instruments, however, are closer to one microphone than to the other; therefore, the sound arrives first at the near mic. The brain uses the arrival-time differences to calculate spatial location.

One key to good spaced-pair recordings is finding the right balance between the stereo spread and the music as a whole. For example, to record a single instrument, the appropriate spread may be as little as 2 feet. To capture the full blend of sound from a large ensemble, on the other hand, you may need to place the microphones 12 or more feet apart.

The main drawback of spaced pairs is the diffuse pickup of off-center instruments. Therefore, it's sometimes a good idea to include a third microphone at the center point (especially when using widely spaced pairs) and mix the resulting signal to both channels. Then again, that same diffuse quality can lend a soft, desirable warmth to the natural reverberant qualities of the space. Your approach depends largely on what kind of sound you're after. Experiment with different mics as well as different spacings. If the

FIG. 6: ORTF, a popular near-coincident miking technique designed to mimic human hearing, specifies two cardioid mics 110 degrees apart with 6.69 inches between the capsules.

Double Your Pleasure

mics are capturing too much room reverb, for example, try using directional rather than omnidirectional mics.

When positioning spaced pairs, it's helpful to heed the 3-to-1 rule, which states that the distance between the mics should be at least three times the distance between the mics and the source. (For more information, see "Recording Musician: Avoiding Phase Cancellation" in the July 1997 issue of **EM**.) In practice, though, following the 3-to-1 rule sometimes results in too large a "hole" in the center of the image, especially with widely spaced pairs aimed at large

ensembles. Mic positioning is therefore critical. Ideally you should audition several setups, listen carefully, and compare the results to determine the optimal arrangement. Move the mics in small increments, because a relatively minor change can make a big difference in sound. Some phase cancellation is unavoidable, but when you find the optimal spacing between the mics, as well as the right balance of direct and reverberant sound, spaced pairs can create exceptional spaciousness and realism.

DECCA TREE

A useful variation on the spaced pair is the Decca Tree, which is essentially a spaced pair with a center mic for filling the hole in the center of the image. Engineers Arthur Haddy and Kenneth Wilkinson, among others, developed this array in 1954 while employed by the Decca Recording Company. In Haddy and Wilkinson's setup, the spaced-pair mics were placed 4.42 feet apart. The center mic was situated
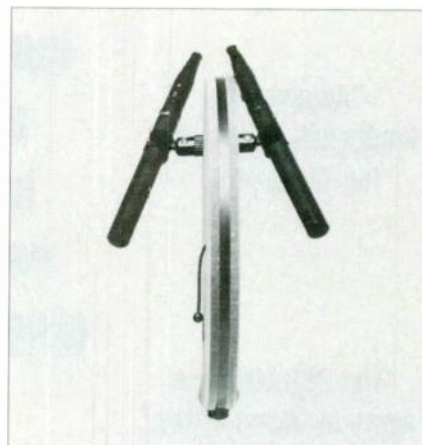
FIG. 8: MB Electronics' Jecklin disk, made in the 1980s, is 0.36 inches thick and covered with foam on both sides. But the foam is thin, so the disk probably reflects too much high end. The cord at the center is for measuring the distance between the capsules and the disk.

between and 2.17 feet in front of the spaced pair. At the time, Haddy and Wilkinson were using Neumann KM 56

BACKGROUND ON BINAURAL

The first experiments with binaural principles were conducted in 1881 by Clement Ader at the Paris Exposition and Opera. Ader positioned a series of carbon microphone pairs (actually telephone elements) about 7 inches apart along the edge of the Paris Opera's stage. The opera performances were picked up by the mic pairs and sent on twin telephone lines to the few subscribers in France that had two lines installed in their homes. The earpiece of one line was held to the left ear, the earpiece of the other to the right. Ader said, "This 'double listening' to sound produces the same effects on the ear that the stereoscope produces on the eye."

Further experiments were sporadic over the next 60 years or so. In the mid-1920s, a few radio stations in the United States began broadcasting on two different frequencies, feeding each transmitter separately from a left-ear and right-ear mic in a dummy head in the studio (see **Fig. A**). Listeners were already using headsets, for the most part, so for this new type of broadcast they just used two radios, putting one mono headset (tuned to the left-ear

station) to one ear and the other mono headset (tuned to the right-ear station) to the other.

By the 1950s, stereo and binaural techniques were often assumed to be the same. To clarify the terms, RCA Victor started including an explanation in the liner notes of all its 2-track stereo open-reel tape recordings. The notes read: "Stereophonic recording differs from Binaural (a term sometimes incorrectly applied to stereophonic records) in that the microphone placements are selected for loudspeaker reproduction. Binaural properly applies to a two-channel system designed for headphone reproduction. It thus requires the use of two channels fed by microphones spaced about seven inches apart (normal ear separation)."

Some West German radio stations devoted time to special binaural transmissions, often of radio dramas called *Hörspiele*. Also, *The Cabinet of Dr. Fritz* series of binaural radio dramas from ZBS Productions was broadcast in the 1980s and early '90s by public radio stations here in the United States. Many of those same sta-

tions also carried binaural expert John Sunier's weekly program, *Audiophile Audition*, which featured his "All Binaural Special" broadcasts four times a year for more than 13 years.
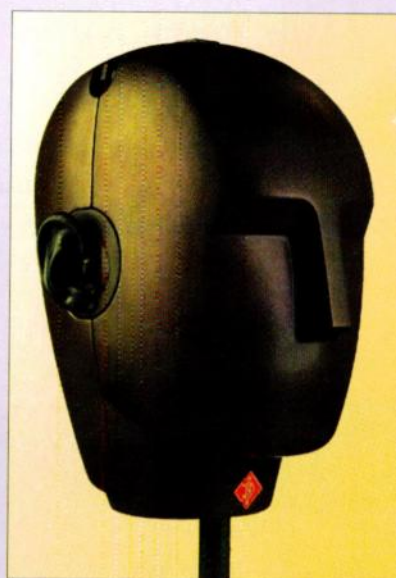
FIG. A: Commercial binaural recordings are often made using dummy heads such as the one in the Neumann KU 100 miking system ($6,995).

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nickel-diaphragm multipattern mics set to omni; later they changed to Neumann M 50 aluminum-diaphragm omnis.

The Decca Tree (see **Fig. 10**) was designed specifically for making stereo recordings of large orchestras. It usually consists of three omnidirectional microphones, although cardioids can also be used. Because the center mic sits in front of the spaced pair, the center signal in a Decca Tree is typically very robust. This can result in comb-filtering effects, so listen carefully to determine optimal placement.

For even larger orchestras, two additional microphones (making a total of five) are sometimes employed as "outriggers" on the sides of the "tree." These can be positioned anywhere from the outer edges of the sound source to approximately one-third of the hall's width in from the side of the hall. Each signal is mixed to the mic channel on its particular side. Decca Trees are generally mounted around 10 to 12 feet in the air and just behind the space where the conductor's head would be.
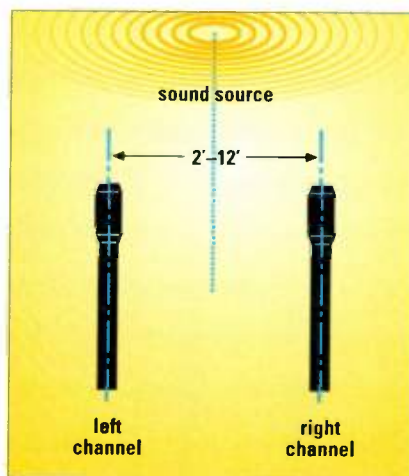
FIG. 9: In a spaced, or AB, stereo pair, two matched microphones are placed several feet apart, parallel to one another and facing the sound source. Stereo positioning (localization) is created by time delay only.

DOUBLE YOUR FUN

I hope I've provided a useful survey of stereo-recording techniques and given you enough information to get you started on some stereo recordings of your own. Perhaps you have a recording project that could benefit from stereo-miking; you should now be able to analyze the situation and choose the most appropriate technique.

However, don't let the techniques limit you. Instead, experiment with them, try variations, and keep notes on the results—just in case you stumble onto something really good. Remember that thousands of talented sound engineers throughout the years have scratched their heads over countless strange acoustic dilemmas, eventually finding solutions through experimentation. Indeed, each of the methods described here was developed under just such circumstances. Who knows—maybe your experiments will lead to the development of a hot new stereo-recording technique.

Elizabeth Papapetrou *has been recording music and writing for music magazines for 17*
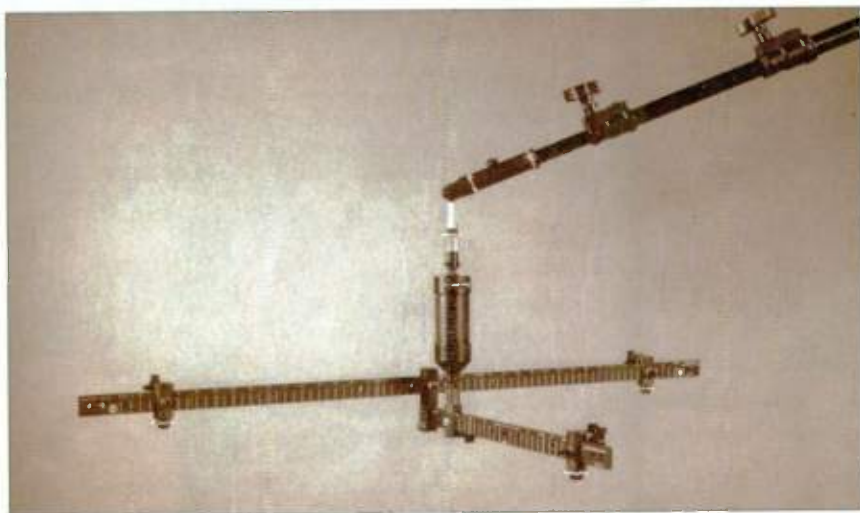
FIG. 10: The Decca Tree array (which consists of a spaced pair supplemented by a center mic) is typically used to record symphony orchestras and other large ensembles. Shown here is AEA's Mini Decca Tree Stereo Microphone Positioner Bar, which Ron Streicher developed to facilitate Decca Tree recordings of chamber groups, jazz bands, and other smaller-scale ensembles.

years. She loves microphones and specializes in recording solo and small-ensemble acoustic music. She is a performing singer, guitarist, and songwriter. Originally from the United Kingdom, she is now based in Gainesville, Florida. Special thanks to Bruce Bartlett of

Crown International (www.crownaudio .com), Wes Dooley and Ron Streicher of Audio Engineering Associates (www.wesdooley .com), David Josephson of Josephson Engineering (www.josephson.com), and John Sunier (www.binaural.com).

here was a time when roll-around office chairs were essential for a well-equipped MIDI studio. Scooting about on four wheels was the only way to reach equipment that needed adjusting, short of standing up and actually walking over to it. Today, thanks to MIDI, digital audio sequencers, and an odd assortment of peripheral gear, you can control an entire studio while seated in one spot. Four-wheel drive is no longer required.

But setting up a central command station for your studio can be daunting. Without a clear understanding of how to configure the software and hardware components of such a system, it's easy to get frustrated and give up before fully realizing the power that you can have literally at your fingertips. In this article I will walk you through the basics of putting together a comprehensive centralized control setup, including hardware, software, and all the dirty details necessary to get things working properly. So strap yourself in, bolt down your chair, and prepare for takeoff.
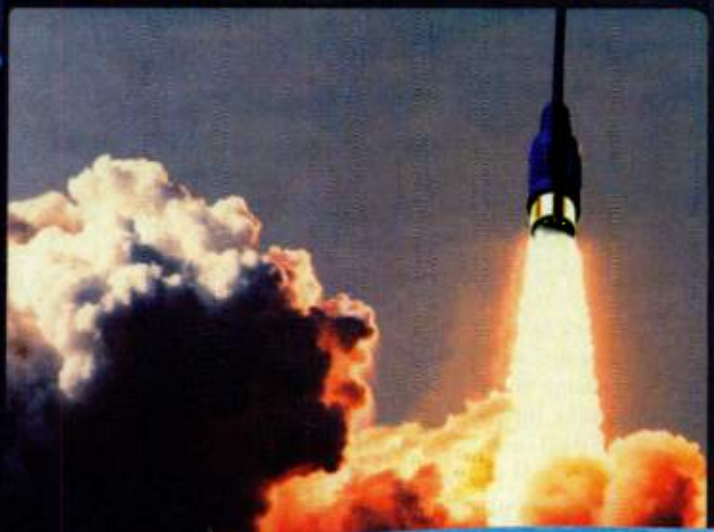
CONTROLLING YOUR MIDI STUDIO FROM A CENTRAL LOCATION.

MISSION CONTROL

BY ERIK HAWKINS

ILLUSTRATION BY DENNIS IRWIN

VSR-880 24-bit DIGITAL STUDIO RECORDER

STATUS/LOCATOR/SCENE /CH EDIT

1 2 3 4 5 6 7 8

EZ ROUTING

SCRUB   V.PITCH   TAP   LOOP   A.PUNCH   ◄ GRADATION ►   SNAPSHOT   AUTO MIX
                                          SCENE LOCATOR CLEAR

Assign   V.Track   Lim E   Pan   EQ   Aux Send   EFFECT 1   EFFECT 2   ROUTING

PREVIEW
→ TO   ↔ THRU   ↦ FROM

ZERO   REW   FF   STOP   PLAY   REC
|◄◄   ◄◄   ►►   ■   ►   ●

STORE   SONG TOP   SONG END   SHUT/EJECT   RESTART

INPUT (TRS)
A
B

PHONES

Roland

MISSION CONTROL

MIDI BRIEF

MIDI was originally conceived in the early '80s as a way for electronic musical instruments to communicate, but it has far outgrown its acronym. Today, almost every piece of studio equipment—not just musical instruments—is outfitted with MIDI, from effects processors to sound cards, multitrack recorders, and even mixing boards.

Written into the MIDI protocol are several data types that allow external control of your MIDI devices. Without them, it would be impossible to hook up a central command station to control your studio. The most important of these data types for our purposes are Control Change (CC), SysEx (including MIDI Machine Control), and Program Change messages.

The MIDI specification defines 128 CC messages (numbered 0 to 127). Almost all gear transmits and responds to at least a few of these—and some to all 128. CC messages handle a wide variety of MIDI parameters including Volume, Pan, and Modulation. (See the table "MIDI Control Change Numbers and Their Associated Data Types" for a complete list of Control Changes.) Many messages are assigned to control specific parameters; others are left open or "undefined," allowing developers to cook up their own assignments.

Although CC messages are generic and apply to all MIDI gear, SysEx messages are product specific. With SysEx, a manufacturer can conjure up a control number for any parameter on its equipment, no matter how esoteric that parameter is. For example, an effects unit's highpass filter frequency-select parameter can be given a specific SysEx value. Control of that particular parameter is then achieved by sending the correct SysEx number to the effects unit—simple. Most MIDI devices have SysEx numbers assigned to all of their parameters, but finding out what these numbers are can be difficult. If it isn't in the manual and you can't access the parameter with a physical control on the unit's face, you'll need to get the information from a user group or directly

from the manufacturer. (Most modern gear, such as Clavia's Nord Lead or Electrix's Warp Factory, transmits SysEx via knobs, buttons, and sliders.)

Program Change messages are nothing new to most MIDI initiates. Send your sound module or effects unit a Program Change, and patches are magically recalled—no in-depth explanation needed here. All MIDI gear capable of storing patches responds to Program Change messages.

A control protocol that was added later in MIDI's history is MMC. (MMC actually employs SysEx messages, but because it's used only for a particular set of operations, simply treating it as its own controller type is acceptable for clarity's sake.) Designed for the remote operation of video and audio decks, MMC is perfect for controlling your MIDI-equipped multitrack (for example, E-mu's Darwin, Roland's VS-880, or an ADAT with a proper MIDI interface). With MMC, you can handle transport functions, enable tracks for recording, set locate points, and so on, so that running your multitrack from afar is a snap. Most professional-level digital audio sequencers support MMC, as do many of today's MIDI digital mixers (such as Tascam's TMD-1000 and Ramsa's WR-DA7). You'll typically find dedicated controls, including transport and location keys, directly on the mixer's face.

BRAIN AND BACKBONE

The cornerstone of a central command station is your digital audio sequencer. It functions as the brain of the system, allowing you to assign, route, and manipulate a wide range of MIDI control data. Nearly all major digital audio sequencers will record and play back CC and SysEx data and, of course, transmit Program Change messages. However, not every program offers comprehensive data remapping (turning one type of control into another, for example) and custom control panel options (or "templates," which I'll discuss shortly). The digital audio sequencers that I worked with when writing this article are all up to the task. This group, consisting of Mark of the Unicorn's *Digital Performer,* Emagic's *Logic Audio,* Steinberg's *Cubase VST,* and Cakewalk's *Pro Audio* 9, is by no means definitive—other worthy programs are certainly available. These are merely the sequencers that I had time to test.

MISSION CONTROL

Although it's possible to control some of your studio using only a digital mixer with MIDI capabilities, you can't record MIDI tracks or automate MIDI performance data without a sequencer. The limitations are obvious (okay, I suppose if live recording is your sole objective, you could get by). In addition, many digital mixers have only onboard, scene-based automation and actually require a sequencer for full dynamic automation. But team a digital mixer with a digital audio sequencer and you've got an extremely powerful combo. Digital mixers make great control surfaces when hooked up to a properly configured controller template (again, more on this later).

The backbone of your command station is the MIDI interface/patch bay. From this device, all MIDI messages coming in and out of your digital audio sequencer are piped to their assigned destinations. Not having a MIDI patch bay is like having a head without a spinal cord: you can see your body, but you can't communicate with it. These days, MIDI interface/patch bays perform a variety of additional chores, including MIDI processing and mapping, data merging, and synchronization; an example is MOTU's MTP AV. A single-port interface without a patch bay is fine for rudimentary systems (a computer and a keyboard, for instance), but for full-fledged control of larger systems, a multiport MIDI interface—which can provide up to 128 MIDI channels—is essential.

TEMPLATE TIPS

Templates are groups of virtual elements (including knobs, buttons, sliders, dials, and text boxes) strung together to make an onscreen MIDI control console (see **Fig. 1**). Many digital audio sequencers (such as *Pro Audio* 9) allow you to build these consoles completely from scratch and to assemble whatever elements are desired for your custom control template. Most programs also offer numerous preset templates for remote control of a variety of gear ranging from generic General MIDI sound modules to specific synth consoles and effects units. Many manufacturers and user groups offer downloadable preset templates in various formats on their Web sites.

How a custom template is created varies from program to program. Generally, it's a matter of adding elements to your virtual console by either choosing them from a menu or doing the click-and-drag dance with a palette bar. You then assign a specific type of MIDI control data, the desired MIDI channel, and a MIDI output port to each element. This is often accomplished by typing information into text fields or by putting the console into "learn mode." Learn mode is the generic term for a ready state in which an element can be taught what type of data it will output. With learn mode on and the element selected, simply send the MIDI data that you want assigned to it (perhaps by moving a knob or slider on a physical hardware device) and you're done. With each element in your console outputting specific control information, you're ready to take command of your targeted MIDI device (which could be an external MIDI sound module, an internal sound card, or even a software synthesizer or sampler).

A well-designed digital audio sequencer will let a

FIG. 2: *Cubase's* **Mixer Maps display the main control functions of your hardware. Here, the basic components of a Kurzweil K2000 are shown.**

virtual console element respond to one type of MIDI data and output a different type. This would, for example, allow a knob or slider on one device to generate CC data with which it can control a parameter on another device that normally responds only to SysEx. The trick in setting up this type of remapping is to be sure that each console element is configured properly, that input and output data fields are assigned correctly, and that your digital audio sequencer's patch-through parameter is enabled. With patch-through on, incoming MIDI data is automatically routed to the control-template element and then out of the MIDI interface to its associated hardware device. (You will often need to select either an instrument track or a dedicated template track to route the incoming data properly.) This is the same method that you would use to route a control keyboard to different devices. The advantage of this system is that any MIDI unit can act as a control surface, whether it's a synth module with knobs that transmit CC data, a dedicated MIDI fader box, or a digital mixer.

Finally, just to clear up any confusion, two common types of templates are available—MIDI templates and audio templates. MIDI templates control internal and external MIDI devices, using either SysEx or CC data. Audio templates typically receive SysEx data generated by external MIDI control surfaces and then route the data to the sequencer's digital audio mixer. Although MIDI templates are completely user-definable, audio templates are not.

FIG. 1: Cakewalk *Pro Audio* **9 provides tools for building custom templates to be used with external MIDI and audio gear. The panel shown here is a preset template for use with the Line 6 Pod.**

MISSION CONTROL

A manufacturer must create a special template or "profile" to allow a control surface, such as Mackie's HUI (Human User Interface) or CM Automation's MotorMix, to control the sequencer's audio mixer section. With luck, users will have the ability to create their own audio templates in the near future.

BREAKING IT DOWN

Five basic groups of gear can be controlled via MIDI: sound modules (which include MIDI keyboards used as sound modules), effects processors, multitracks, mixers, and audio control devices (for example, VCA automation devices—more on these soon). We'll look at all of these groups in turn. Each has a set of parameters that is usually remote-controlled (see the table "Typical Remote-Control Parameters and Their Control Types").

I'm probably stating the obvious, but it's important to mention: any equipment that can be controlled via MIDI can be automated. And, of course, the automation data (the information that's recorded into your sequencer) can be endlessly tweaked to your heart's delight. With the powerful data-

FIG. 3: Among the various MIDI control surfaces on the market today, the keyboard is often overlooked as a resource. Here, the keyboard sits near the Keyfax PhatBoy and the CM Automation MotorMix. Tape strips are handy reminders of the controller maps I've assigned to the keys.

manipulation tools found in most of today's sequencers, the possibilities for using MIDI to control all types of gear are endless.

Synth-O-Matic. Sending program changes to MIDI sound modules is the inalienable right of all MIDI musicians. Every digital audio sequencer that I've ever encountered can generate Program Change messages, making the remote switching of sounds a snap.

But patch changing only scratches the surface of what you can do with most modern sound modules. A huge range of parameters can be controlled remotely, including volume, envelope times, resonance amounts, and even built-in effects. Generally, these parameters are accessible through Channel Voice or SysEx messages.

With a control template properly configured for your module, sounds can be designed on the fly. So your keyboard player could be practicing a part

COMMANDING PRESENTS

In addition to CC and SysEx messages, a well-implemented MIDI sequencer will provide a host of other MIDI commands that can simplify and enhance your studio setup. Here are a few that make my life easier.

A MIDI keyboard can be used to send Aftertouch for modulating effects. For example, I've found that laying into the keys to speed up or slow down a Leslie effect offers a more tactile approach than using an onscreen slider, a modulation wheel, or even a footpedal.

A sequencer template with Local Control Off messages can eliminate the routine of burrowing through submenus on your keyboard just to

set it up as an effective MIDI controller.

In the old days, layering synth sounds was a matter of selecting complementary sounds on each, setting the desired synths to the same MIDI channel, and perhaps disabling Program Changes so that the proper patches would be retained. Most modern sequencers can assign a single MIDI track's output to a group of devices, each receiving on its own MIDI channel. These virtual synth stacks can be edited, saved, and recalled together with the proper patch for each synth.

Tired of kneeling by your rack just to change your sound module's default mode? Fed up with hearing

your old Casio synth play your drum patterns? Sending Channel Mode messages (for example, Omni Off, Mono) from your sequencer instructs your errant gear to mind its own business.

Is your master MIDI controller a tad weak in the MIDI implementation department? Your sequencer (or even your MIDI patch bay on its own) probably has a slew of remapping capabilities that can send commands that your original controller wasn't designed to send. For example, my MIDI guitar controller doesn't send Aftertouch, but with my sequencer I can easily remap the guitar's expression pedal to send it.

—Marty Cutler

Sonic Irregularities gettin' you down?

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If your mix is in need of a dose of sonic TLC, the amazing dual channel CEL-2 compressor/expander/limiter may be just what the doctor ordered. Three high quality products all in one, the CEL-2 does a thorough but gentle job of soothing sonic irregularities in any recorded or live sound application. And it doesn't even upset your mix. Discrete circuitry allows each of the CEL-2's three sections to be independently operated and controlled. Intuitive and accurate metering makes the unit incredibly easy to use. The awesome speed of its onboard THAT™ processor alone is a prescription for success. The CEL-2 is, hands down, the most professional product in its category. And better still, it costs less than the competition!

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MISSION CONTROL

at the same time you're tweaking that keyboard's current patch from your computer. Even after the part has been recorded, you can continue to adjust parameters while the track is playing back. Best of all, the template on your computer monitor can be as big or as small as you like—no more staring into a tiny, poorly illuminated screen.

Affecting effects. Most of today's effects processors can store patches. As you would expect, these patches can be switched using program changes. This lets you create dramatic shifts in effects over the course of your composition. For example, you could dial up a small-room reverb for your verses to create a tight, intimate sound, and then jump to a large-hall reverb on the choruses for a heavenly cathedral effect. Just be careful when you swap patches; a program change in the middle of a reverb tail can sometimes cause a nasty pop.

An effects processor's sound-sculpting parameters (such as EQ, compression, distortion, and ADSR) can usually be controlled using SysEx or CC data. If you set up the elements in your template to send the proper messages, you'll have command over a myriad of sound-mangling possibilities. My favorite trick is assigning a fader element to a resonance parameter: slide the fader up and down in time with your music while recording the performance to create automated resonant sweeps that match your beats. If your effects unit lacks a resonance parameter, try using a distortion setting or a phaser/flanger value. Automating effects changes is a blast.

Multitrack racket. Often, audio gear makes an awful noise, from whining motors in tape-based machines to whirring fans in hard drive systems. Getting this equipment out of hearing range greatly improves the listening environment of your control room. With MMC you can shove your multitrack into the far corner of the room without giving up one iota of control; you should need to touch the machine only for tape loading or hard drive backup.

An MMC command panel can be either virtual (on your computer screen)

MIDI CONTROL CHANGE NUMBERS AND THEIR ASSOCIATED DATA TYPES

Control Change Number	Data Type	Control Change Number	Data Type
0	Bank Select	64	Damper Pedal On/Off (Sustain)
1	Modulation Wheel	65	Portamento On/Off
2	Breath Control	66	Sustenuto On/Off
3	Undefined	67	Soft Pedal On/Off
4	Foot Controller	68	Legato Footswitch
5	Portamento Time	69	Hold 2
6	Data Entry	70	Sound Controller 1 (Sound Variation)
7	Channel Volume (formerly Main Volume)	71	Sound Controller 2 (Timbre/Harmonic Intensity)
8	Balance	72	Sound Controller 3 (Release Time)
9	Undefined	73	Sound Controller 4 (Attack Time)
10	Pan	74	Sound Controller 5 (Brightness)
11	Expression Controller	75	Sound Controller 6 (Decay Time)
12	Effect Control 1	76	Sound Controller 7 (Vibrato Rate)
13	Effect Control 2	77	Sound Controller 8 (Vibrato Depth)
14	Undefined	78	Sound Controller 9 (Vibrato Delay)
15	Undefined	79	Sound Controller 10
16	General-Purpose Controller 1	80	General-Purpose Controller 5
17	General-Purpose Controller 2	81	General-Purpose Controller 6
18	General-Purpose Controller 3	82	General-Purpose Controller 7
19	General-Purpose Controller 4	83	General-Purpose Controller 8
20	Undefined	84	Portamento Control
21	Undefined	85	Undefined
22	Undefined	86	Undefined
23	Undefined	87	Undefined
24	Undefined	88	Undefined
25	Undefined	89	Undefined
26	Undefined	90	Undefined
27	Undefined	91	Reverb Send Level
28	Undefined	92	Effects 2 Depth (Tremolo Depth)
29	Undefined	93	Chorus Send Level
30	Undefined	94	Effects 4 Depth (Celeste/Detune Depth)
31	Undefined	95	Effects 5 Depth (Phaser Depth)
32	Bank Select	96	Data Entry +1
33	Modulation Wheel	97	Data Entry −1
34	Breath Control	98	Nonregistered Parameter Number LSB
35	Undefined	99	Nonregistered Parameter Number MSB
36	Foot Controller	100	Registered Parameter Number LSB
37	Portamento Time	101	Registered Parameter Number MSB
38	Data Entry	102	Undefined
39	Channel Volume (formerly Main Volume)	103	Undefined
40	Balance	104	Undefined
41	Undefined	105	Undefined
42	Pan	106	Undefined
43	Expression Controller	107	Undefined
44	Effect Control 1	108	Undefined
45	Effect Control 2	109	Undefined
46	Undefined	110	Undefined
47	Undefined	111	Undefined
48	General-Purpose Controller 1	112	Undefined
49	General-Purpose Controller 2	113	Undefined
50	General-Purpose Controller 3	114	Undefined
51	General-Purpose Controller 4	115	Undefined
52	Undefined	116	Undefined
53	Undefined	117	Undefined
54	Undefined	118	Undefined
55	Undefined	119	Undefined
56	Undefined	120	All Sound Off
57	Undefined	121	Reset All Controllers
58	Undefined	122	Local Control On/Off
59	Undefined	123	All Notes Off
60	Undefined	124	Omni Mode Off (+ All Notes Off)
61	Undefined	125	Omni Mode On (+ All Notes Off)
62	Undefined	126	Poly Mode On/Off (+ All Notes Off)
63	Undefined	127	Poly Mode On

or real world (as on a MIDI digital mixer). A typical MMC panel in a digital audio sequencer is shown in **Fig. 2**. Accessing your multitrack functions on-screen is okay, assuming that you don't mind punching in with your QWERTY keyboard or rewinding with your mouse. I much prefer the feel of physical keys or buttons—such as those found on a digital mixer—or a device like the JLCooper Cue Point. These controls are much more ergonomic and reassuring, giving you the feeling that the multitrack is right under your fingertips.

The main thing to watch out for when configuring an MMC system is closed-loop versus open-loop operation. Choosing closed- or open-loop designates the type of communication to be used between the controller and the multitrack (specifically, whether two-way communication is required). Both the digital audio sequencer and the multitrack should be assigned the same setting; otherwise, you're asking for problems. Check the manuals to see if your multitrack and sequencer are compatible.

MMC has two small drawbacks. First, it's using MIDI; there's some delay between the time you hit an MMC button and get a response from the multitrack. This isn't usually a problem, but when you need a tight punch, it can be unac-

FIG. 4: Teaming a physical control surface, a digital audio sequencer, and an audio patch bay makes for a very strong combo. The author's mixing station, complete with a HUI, Pro Tools, and a PM-216 patch bay, is shown here.

ceptable. Every system has a different amount of machine lag, and there's no telling how bad yours might be. For razor-sharp punches, however, try automating the record-in and record-out points. This usually does the job, because the digital audio sequencer compensates for time lag to make the perfect punch. MMC's second drawback is that its panels don't mirror your deck's meters; this requires that you keep your multitracks in view for monitoring.

Digital mixers. Most digital mixers can do more than just control your audio—they can also control your MIDI tracks. A good digital mixer will have a

mode dedicated exclusively to handling MIDI. In that mode, the console's input strips become MIDI control strips, with each one corresponding to one or more MIDI tracks in your sequencer. By flipping between the board's two modes, you're able to command both audio and MIDI from a single control surface.

At the most basic level, you'll have control over track volume, mute, and pan. This is usually done by sending Control Change data from the mixer's physical controls. You may find that the mixer allows you to send other forms of data too, such as SysEx messages. As usual, any or all of this information can be mapped to a template of your creation.

Keep in mind, however, that the same control on each channel strip typically transmits identical forms of data—for example, every fader transmits CC 7. This is why each strip must have its own MIDI channel; otherwise, one-to-one discrete control of individual MIDI tracks would be impossible. (On certain consoles, the MIDI channels and control types can be flexibly assigned, but this is the exception, not the rule—normally, consoles are preset.)

Another use for a mixer is to control the individual sounds of a multitimbral module, especially one with just stereo outputs (as is the case with many GM modules). Using the mixer's audio mode, set the module's stereo output to a comfortable level (0 dB, for example)

TYPICAL REMOTE-CONTROL PARAMETERS AND THEIR CONTROL TYPES

Device	Parameter	Control Type
Sound Module	Patch	Program Change
	Real-time parameter (for example, Volume, Pan, ADSR)	Control Change and SysEx
Effects Module	Patch	Program Change
	Real-time effect (for example, EQ, resonance, delay)	Control Change and SysEx
Multitrack	Transport	MMC
	Record enable	MMC
	Locate	MMC
Digital Mixer	Volume	Control Change
	Mute	Control Change
	EQ	SysEx
	Scene	Program Change
Audio Control Device	Volume	Control Change
	Audio patch bay	Program Change

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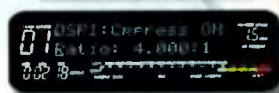
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ALESIS

MISSION CONTROL

and lock it in. Then, when you want to change a specific patch's level on the module, simply switch to the MIDI mode and grab the appropriate fader. If you have all of your MIDI assignments and templates configured correctly, controlling levels in this fashion should be a piece of cake.

Audio control devices. A number of MIDI peripheral devices for controlling audio are now on the market. They range from voltage-controlled amplifiers (VCAs) used for handling signal levels to MIDI-automated audio patch bays and remote-controlled preamps. If you're missing that final link in your quest for ultimate studio control, this diverse group of gear may be the key.

VCA modules (such as the Niche Audio Control Module) are great for controlling volume levels on a multitrack, especially if you don't have an automated mixer. All VCA units come with an audio input and output. Insert the unit between the output of your multitrack and the input of your mixer. Make the necessary MIDI connections and set up a template for control. Most VCAs use a single MIDI channel and respond to consecutive CC numbers (one number per audio channel). Using your template's faders, you can now automate the levels of your multitrack—pretty cool.

MIDI-controlled audio patch bays are rare but handy. A 16-by-16-point unit called the AudioMatrix Patcher was once made by 360 Systems. Currently in production is CM Automation's 16-by-16 PM-216; the company also makes a 32-by-32 model, the PM-64. You can control the PM-216's routings by software (Windows only) for comprehensive remote patching. Patches can be saved into one of 64 program slots. This means that—you guessed it—the programs can be recalled using Program Change messages. It's a perfect patch bay if you have, say, only a 12-channel mixer

and 16 sound sources. Using automated patch changes, you could actually switch the inputs to the mixer while your song is playing, without missing a beat. (Of course, in this example you're limited to just 12 inputs, but you get the idea.)

Who knows what the manufacturers will cook up next to satiate our audio-control needs? Summit Audio's new *Extension 78* plug-in for Digidesign's *Pro Tools* 5.0 is probably a good clue. This plug-in gives you direct MIDI control over Summit Audio's Element 78 line of Class A analog signal processors and preamps. No template programming here—just make the MIDI connections, call up the plug-in, and sit back: you've now got complete control of external equipment directly from your digital audio sequencer. (For those who haven't heard, *Pro Tools* 5.0 now has a MIDI sequencer, so it's officially a digital audio sequencer.) Imagine the possibilities—I'm sure we'll be seeing many more hardware/plug-in systems like this for every conceivable application.

CONTROL ISSUES

Many fine real-world control surfaces with physical dials, keys, and sliders to twist and push are currently available. They range from a studio keyboard to a module with SysEx buttons to a box with knobs that generate CC data. Anything that transmits MIDI messages can be mapped in a control template.

A control surface that's rarely thought of as such is right under our noses— the common MIDI keyboard. All keyboards have a modulation wheel that's perfect for controlling volume. Just create a template, map CC 1 (the mod wheel's controller number) to output as CC 7, pick a device to control, and voilà—a nifty level controller. This same concept can be applied to any of the keyboard's buttons (assuming that

MIDI CONTROL SURFACES		
Product	**Manufacturer**	**EM Review**
CS-10	JLCooper	not reviewed
HUI	Mackie	January 2000
MCS-3000	JLCooper	not reviewed
MotorMix	CM Automation	March 2000
PC 1600x	Peavey	March 1993
PhatBoy	Keyfax	January 1999
Regalwerk	Doepfer	not reviewed
StudioMix	Peavey	November 1999

MISSION CONTROL

they transmit their values via MIDI) and even to the ivories (see **Fig. 3**). If you don't have a keyboard but you own some other type of MIDI instrument—drum pads, guitar controller, or wind instrument for instance —there's no reason that you can't use it in place of a MIDI keyboard.

Many types of dedicated controller boxes have recently shown up on the scene (see the table "MIDI Control Surfaces"). The simplest of these boxes are diminutive in size, fitting neatly on a desktop or the face of a keyboard, and are reasonably priced (less than $500). They're outfitted with some combination of faders, buttons, and knobs that transmit either CC or SysEx messages. The MIDI controller data that they output is usually fixed, but some units contain patches or different modes with remapped sets of controller data. Usually, the patch parameters will match a specific piece of hardware or software (for example, the PhatBoy has a mode to control Roland's JV-1010/2080).

The most complex breed of MIDI controllers is the mixer control surface, such as the MotorMix or HUI (see **Fig. 4**). These units have their own proprietary sets of SysEx messages that must be mapped by the digital audio sequencer's manufacturer to work with the program's audio tracks. Although it is theoretically possible to map mixer control surfaces' SysEx messages to a MIDI control template, you certainly have better things to do with your time, such as making music. These devices are fabulous for controlling the audio in your digital audio sequencer, whether you're writing complex automation moves or tweaking plug-ins—but unless preset MIDI control templates are included, forget about using them as MIDI track controllers.

CONTROL FREAK

No matter how your studio is laid out, stacked up, or spread around, the bottom line is that it can all be controlled from one central location. This is a definite ergonomic advantage, regardless of your situation. But even if having complete control over every nook and cranny isn't your cup of tea (because, honestly, setting up all of this stuff can take quite some time), having remote control over at least a few of your favorite parameters is wonderfully convenient. Being able to switch patches on your sound module, control your multitrack, design a new effect, or automate an old one—all without leaving your computer—can, in the long run, be a great time-saver.

In the end, what's important is not that we have every item in our studio hooked up to and controlled by our central command station, but that we have the *ability* to make it so. I hope that this article has painted a picture of the possibilities. A little tweaking can go a long way toward making your studio much more efficient and powerful.

Erik Hawkins is a musician and producer working in Los Angeles County and the San Francisco Bay Area. Visit him at www .erikhawkins.com for more equipment chitchat and tips on what's hot for the personal studio.

Recording Artists

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Desktop

Spice up your next project with a bit of Spanish flavor.

If you've never seen a full-blown flamenco production, you're really missing something: brightly costumed dancers combining fluid movements with dazzling footwork; flashing fans and chattering castanets; clapping, yelling, and stomping—the excitement and spontaneity are positively riveting. And energizing it all is music so lively and compelling, you'll find yourself on the edge of your seat. Whether it's the mournful, heartfelt laments of the *seguiriyas;* the driving, pulse-pounding rhythms of the *bulerías;* or the lively, festive sounds of the *alegrías;* flamenco has something for everyone.

It is therefore not surprising that flamenco music has enjoyed a kind of renaissance lately. Forward-thinking guitarists such as Paco de Lucia, Vicente Amigo, and Tomatito have introduced flamenco guitar techniques, styles, and idioms into other types of music, and they have explored the use of nontraditional instruments within the context of flamenco. Groups like the Gipsy Kings have gained enormous popularity by adopting a pop-music style based largely on a few simple flamenco techniques and rhythms (mainly *rumbas*).

Film scoring is one field in which knowledge of flamenco might serve you well. Movies such as the recent *Mask of Zorro* (starring Antonio Banderas) provide many opportunities to flavor a soundtrack with some Spanish-sounding music. In fact, flamenco music is featured in a number of movie classics.

BY DAVID RUBIN

That's Vicente Gomez, for example, playing the guitar parts in the Tyrone Power movies *Blood and Sand* and *Captain from Castile.* (If you'd like to see motion pictures based entirely on real flamenco performances, check out the films of director Carlos Saura, especially *Carmen*—a mesmerizing flamenco interpretation of the Bizet opera.)

flamenco

Desktop Flamenco

A little flamenco music can go a long way in scoring for television commercials as well. For example, Los Angeles–based studio guitarist Ted Owens was recently called in to lay down a flamenco guitar track for a Nissan commercial. The spot called for several seconds of upbeat flamenco playing that segued into a rock piece. Fortunately, Ted is a versatile guitarist who can play in any style. He cooked up a convincing *bulerías* that led into the rock section, and the producers went away happy.

But what if you aren't a guitarist, and

Yolanda Arroyo (in red) surrounded by her dance students at a recent performance.

you find yourself in a similar situation? In this article, I'll discuss some of the desktop music tools that can help you add the excitement of flamenco to your next project. With a sampler, a sound module, and a MIDI

A BRIEF HISTORY OF FLAMENCO

To fully appreciate flamenco, it's important to know a little about its origins. The term *flamenco* refers to a genre of Gypsy songs, instrumental music, and dance that evolved primarily in a region of southern Spain called Andalusia. The Andalusian provinces include such familiar-sounding cities as Seville, Granada, and Málaga, which is why many flamenco pieces incorporate those names.

During the Middle Ages, the Moors invaded the Iberian Peninsula and succeeded in driving the Spanish aristocracy into the northern parts of the country. For several hundred years thereafter, Andalusia flourished as a trading center and cultural melting pot, giving rise to great works of art and architectural marvels such as the beautiful Alhambra and the Alcazar, still popular with tourists today.

It is generally believed that the Andalusian Gypsies originated in India (probably in the Punjab region) and later migrated north to escape persecution. During the early 15th century, many found their way to southern Spain, where they encountered a rich variety of religious and secular musical styles from an array of Eastern and Western cultures. Unfortunately for

the Gypsies, during the latter half of the 15th century, the Spanish aristocrats gained sufficient power to reclaim the country. In 1492—as Christopher Columbus waited impatiently for the go-ahead from Queen Isabella—the Spanish kings expelled the last of the Moors from Granada and reestablished the country under a unified Spanish rule.

By the early 16th century, the political climate had clearly changed under the Spanish monarchs. Gypsies, Jews, and Arabs were actively persecuted by the Christian rulers. As a result, these seemingly disparate "outlawed" groups banded together against their common oppressor; in the process, the Gypsies began to incorporate Moorish and Jewish musical elements into their songs. This is hardly surprising, for the Gypsies had always been adept at absorbing and adapting musical elements from different cultures. The resulting amalgam of Indian, Arabic, Hebraic, liturgical Christian, and Andalusian folk musical styles was fused in the crucible of Gypsy experience, ultimately giving birth to the exciting music that we now know as flamenco.

In its early years, flamenco consisted primarily of solo songs with

little or no accompaniment. These songs—which typically touched on the suffering, misfortune, and oppression of an outcast minority—were sung at Gypsy gatherings and in the caves where the Gypsies often lived. Over time, many types of songs were partnered with guitar accompaniments and dancing.

By the mid-19th century, flamenco had grown increasingly popular with non-Gypsy Spaniards and European tourists, and flamenco performers became essential fixtures in numerous restaurants and small cafés throughout Spain. This "Café Cantante" period (1850–1910) is often referred to as the Golden Age of Flamenco; the art form reached its zenith during this time, defining much of what we see and hear in today's flamenco performances.

Ongoing experiments in flamenco continue to push the artistic envelope by combining flamenco's techniques, sounds, and rhythms with jazz, pop, classical, and other musical styles (much to the dismay of flamenco purists). It is not unusual today to find flamenco guitarists and singers collaborating with players of nontraditional instruments such as flute, saxophone, and cello.

keyboard controller, you have the essential tools you need to infuse your music with a dash of exotic spice.

Of course, "desktop" flamenco can never fully capture the depth and spirit of the real thing. But if you understand how the elements of flamenco interact to create that characteristic sound, you can open the door to many new creative possibilities. Unfortunately, sampling CDs and other resources seldom offer any help in this area, so I'll begin by introducing the elements of flamenco music and the roles that they play. A little later I'll look at flamenco sampling CDs from Discovery Firm, Universal Sound Bank, Zero-G, and Best Service.

THE SOUNDS OF FLAMENCO

The three primary elements in a flamenco performance are the song (cante), the dance (baile), and the guitar playing (toque). These can occur in any combination, with performances ranging from intimate solos to large ensemble productions.

Flamenco music can trace its lineage at least as far back as the 15th century, when Moorish, Jewish, Indian, Christian, Byzantine, and regional Spanish music coalesced into the musical art form of the Andalusian Gypsies (Gitanos) in southern Spain. In its early days, flamenco served as the private expression of a long-suffering outlawed people, but by the mid-19th century flamenco had emerged as an international cultural phenomenon. Today, when most people in the United States think of Spanish music, flamenco is what first comes to mind. (For more on the evolution of flamenco, see the sidebar "A Brief History of Flamenco.")

Although its style borrows heavily from the Spanish folk-music tradition, flamenco retains its unique sound due in large part to the musical elements absorbed from Arabic, Jewish, Indian, and other Eastern cultures. Many flamenco songs clearly show these influences, with modal melodies, melismas that seem to go on forever, microtonal passing tones, and other chantlike qualities. However, flamenco also includes many works with simple folk-like chord progressions that are clearly not of Gypsy origin.

Flamenco pieces are named for the song form upon which they are based, although singing is not always involved. For example, names such as *soleares, bulerías, alegrías,* and *tientos* are not specific titles of fixed pieces, but rather musical forms with associated meters, rhythmic patterns, tempos, harmonies, and other characteristics. (More on these later.)

Improvisation and spontaneity are also important elements in flamenco. During a performance, onstage participants add rhythmic clapping (palmas) and shout quips and encouragement (jaleo) to singers and dancers. Audience members, too, shout their approval as the performance gains momentum, and the room acquires the atmosphere of a festive jam session (juerga).

Song. In recent years, dancers and guitarists have often eclipsed the singer (cantaor) in flamenco shows, especially when presented to audiences that are largely non-Spanish-speaking. But during its genesis, flamenco centered entirely on the song, which was offered with only simple rhythmic accompaniment or none at all. (Guitar and dance were not added to flamenco until the 19th century.) The song provided the Gypsies an emotional outlet and a means of expressing the grief and suffering that they endured daily. Com-

The author's wife, "Kata," performs the *sevillanas* at a recent performance in Hollywood.

mon song themes include sickness, persecution, death, prison, and of course, unrequited love.

Good flamenco singers are prized for their coarse, unrefined vocal quality (think the opposite of Bing Crosby); the evocative, wailing, strained delivery of many songs is often difficult for modern audiences to appreciate. However, these songs best illustrate the non-Western influences that have shaped the flamenco sound and offer a direct link to the ancient *gitano* aesthetic. Examples of these ancient song forms include *soleares* and *seguiriyas.*

On the other hand, many song forms in the flamenco repertory (such as *alegrías* and *bulerías*) are more upbeat in nature and typically deal with less profound issues. These and other popular song forms, such as *rumba gitano* and *tangos flamencos* (not to be confused with the Argentine tango), work well in a variety of modern musical settings and are readily appealing, so they're often included on flamenco sampling CDs.

Dance. When dancing was first added to flamenco performances, a clear distinction existed between the male and female styles. The man concentrated on
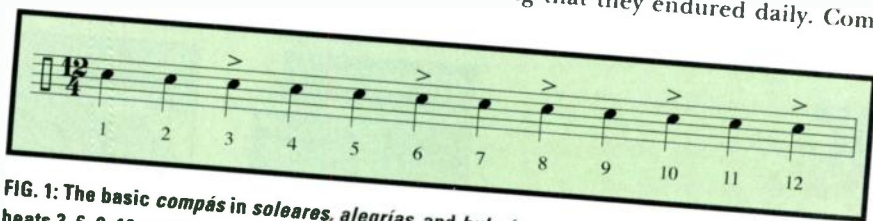
FIG. 1: The basic *compás* in *soleares, alegrías,* and *bulerías* consists of 12 beats with accents on beats 3, 6, 8, 10, and 12.

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Each HDR24/96 track "expands" into eight virtual tracks. Record multiple takes and then comp them into a perfect final. Virtual tracks can also be converted into completely new tracks or divided up into Regions and distributed into existing ones.

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Unlimited cue points with looping and auto-punch-in mode. Cue points are visible on screen or can be accessed from the sidelist that includes History, Groups and Regions.

Track grouping. Lock related tracks (such as drum or background vocals) together and manipulate them as one track.

Quantization. Define a time line and then "snap" your edits, moves and inserts precisely to these points.

Time Bar with user-defined resolution (sample, ¹/₄ frame, msec, second, BBT); Punch, Loop, Cue and Tempo Change markers; Snap function; Zoom arrows; Locate; Punch In/Out, AutoPunch, Open End, Pre-Loop and Post-Loop functions.

Okay, we'll stop with the jargon. You get the idea. The HDR24/96 is a serious digital workstation disguised as a mild-mannered hard disk recorder. With a rich, but non-garish visual interface that you won't mind staring at for hours on end.

You don't need an extra computer. You don't need to spend megabucks. Visit your Mackie dealer for a demonstration of how the HDR24/96 can boost your creativity.

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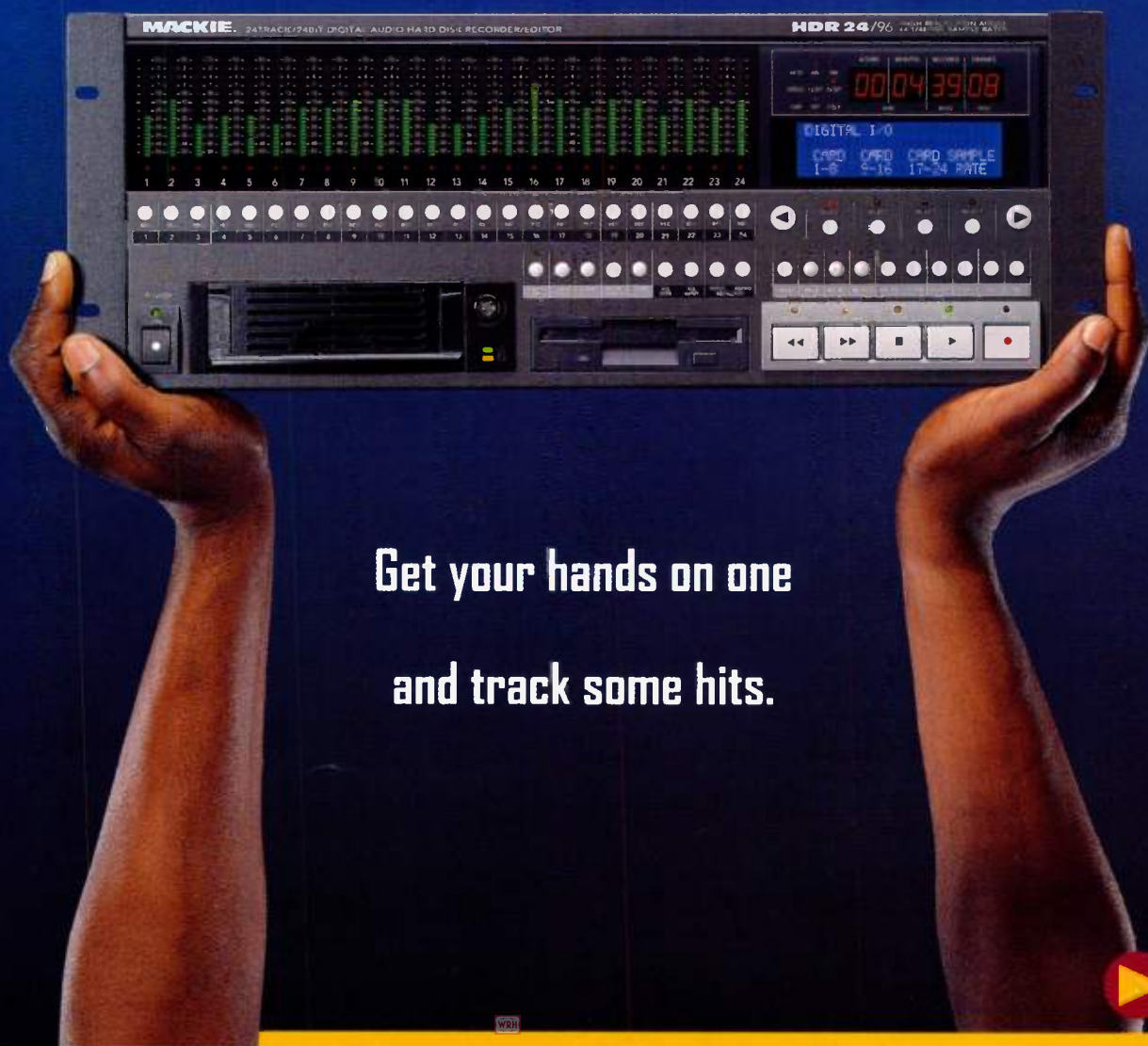
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24x4 LCD display shows operation information, prompts and selections options via paging buttons.

SELECT buttons correspond to options on the screen above them.

+ and – (up and down) buttons scroll through parameter options.

Good 'ole familiar analog-type transport controls.

SYNC selects sync flavor (internal, word clock etc.).

DIGI I/O configures I/O cards.

SYSTEM is the zillion-and-one-options button for accessing preferences and system setup details.

LOCate 1 & 2 buttons select cue or loop points.

Drive bay for Mackie Media™.

(inside) Internal hard drive with recording time equivalent to M•60 Mackie Media.

STORE saves commands.

3.5" disk bay imports tempo maps & upgrades.

LOOP 1-2 selects loop start and end.

RECord SAFE prohibits track arming.

ALL INPUT arms all tracks.

AUTO INPUT—On monitors playback in Play, and Input in Record/Stop/FF/REW for armed tracks. Off monitors input for armed tracks.

DELETE LAST.

TIME CODE chase.

AUTO TAKE.

TRACK selects options for slide and nudge, etc.

PROJECT saves and loads project information.

DISK UTIL & BACKUP control disk operations.

MSRPs, ESPs and TtTESPs*.

FIGURING THE TRUE COST OF A HARD DISK RECORDER.

As with cars, the base cost of a hard disk recorder represents a *starting* point for comparison.

But it doesn't reflect two critical pieces of information: *1)* what you're going to *really* pay to get it up and working in your studio and; *2)* the long term cost of ownership. Consider these factors when pricing hard disk systems…

I/O cards. You're going to need three 8-channel I/O cards to interface a hard disk recorder with your analog or digital console. Mackie's full line of I/O cards starts at just $99 each.

Media recording capacity. The HDR24/96 ships with over NINETY MINUTES of 24-channel recording capacity. This is a major consideration: some recorders ship with as little as 45 minutes of recording media.

Compatibility and connectivity. Our recorder's built-in Ethernet "HDR Bridge" lets you FTP data from the recorder to your computer's desktop, existing SCSI

*Tony the Transhipper's Extra Special Price with Bolivian Warranty

disks and the outside world at near realtime speed (100kBs/second).

The cost of digital editing. Our hard disk recorder is a lot more than just a box that records. It's your gateway to creative non-linear editing without having to lay out big bucks for a full-blown workstation. Mackie's HDR24/96 is an *integrated system* that requires as little as $300 worth of standard PC peripherals to become a $10,000 workstation. Some recorders require adding a complete new computer system (with the interfacing issues that inevitably result).

Long term cost of ownership. What's it going to cost to run the recorder through project after project? Mackie's M•60 media are significantly less expensive than our competition's external SCSI pull-out disks. And they eliminate the need for extra backup devices like Jaz or M/O drives.

Do your homework. Then decide. We think you'll see that, as usual, Mackie offers the best combination of value and performance. Experience the HDR24/96 at a Mackie dealer soon.

Right out of the box, you can start using the HDR24/96 just by hitting REC + PLAY.

You can instantly punch-in as many as 24 tracks at once without the momentary delay that plagues many computer-based recording systems.

You can turn it into a fullblown digital audio workstation just by plugging in a monitor, keyboard and mouse.

You can use it with any mixer, not just with the Digital 8•Bus.

It uses media so affordable that you can keep one around for each project and treat 'em just like tapes[1]. And so rugged that they can withstand heavy impacts that would doom a creampuff SCSI drive.

The HDR24/96 syncs with video, SMPTE and MIDI so you can quickly integrate it with existing post production workstations.

It's sonically impeccable with 144dB dynamic range[2], 0.00001% THD and 2Hz to 22kHz response ±0.5dB[3].

[1] i.e., leave stacks of them out of their cases to gather dust, carry them around in a backpack, use them as doorstops, etc.
[2] Internal processor measurement.

It ships with what we think is the most functional, and best-looking software interface in the pro audio industry. Software so easy to use that HDR24/96 beta testers tell us they don't even have to "crack the manual."

Multiple HDR24/96s can be linked with *sample accuracy*.

Its built-in 100kB/sec Ethernet "HDR Bridge" lets you send files to any computer desktop at blazing speed.

And you have your choice of two remotes: our compact "engineer in a box" Remote 24, or the mondo Remote 48 PRO that runs two HDR24/96s at a time (and impresses the heck out of clients).

In short, the HDR24/96 can turn any console, from an analog 24•8 to a Digital 8•Bus, into a serious hit factory.

And, equally as important, it comes with Mackie's proven track-record for bullet-proof reliability and superb customer support.

[3] At 48kHz sample rate.

We were gonna encrust the HDR24/96's on-screen TOOL panel with our typical tangle of call-out lines and captions until we realized that the darned thing is so intuitive and analog-like (analog-esque?) that it pretty much explains itself.
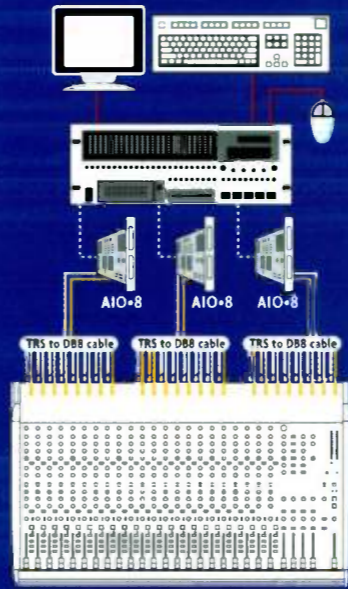
It makes beautiful music with the Digital 8•Bus, too.

The HDR24/96 works great with mixers as small as our analog 24•8. But, as you might imagine, it really comes into its own when you sync it with a Mackie Digital 8•Bus. From the D8B you can arm recorder tracks, operate the transport, select cue points and move or scrub back and forth with that big jog shuttle wheel.
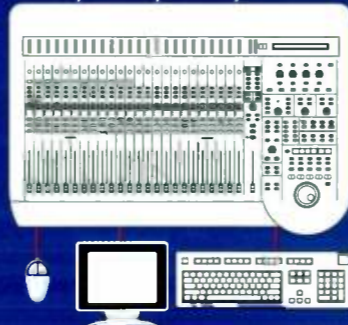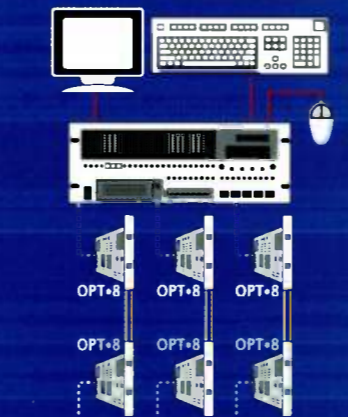
HDR24/96

TYPICAL ANALOG CONSOLE HOOK-UP
TO HDR24/96 VIA AIO•8 I/O CARDS

Sends to recorder can be patched from console bus outs or from channel direct outs (if you have a 24•4 or 32•4, you have at least 24 of them). Returns are patched to channel line inputs. Keyboard, monitor and mouse are not required.

TYPICAL DIGITAL CONSOLE HOOK-UP
TO HDR24/96 VIA OPT•8 I/O CARDS

OPT•8 cards have ADAT optical inputs and outputs. DIO•8 cards (ADAT optical & TDIF with 16-to-24-bit conversion and format conversion) or PDI•O (AES/EBU with sample rate converter) may also be used in any combination.
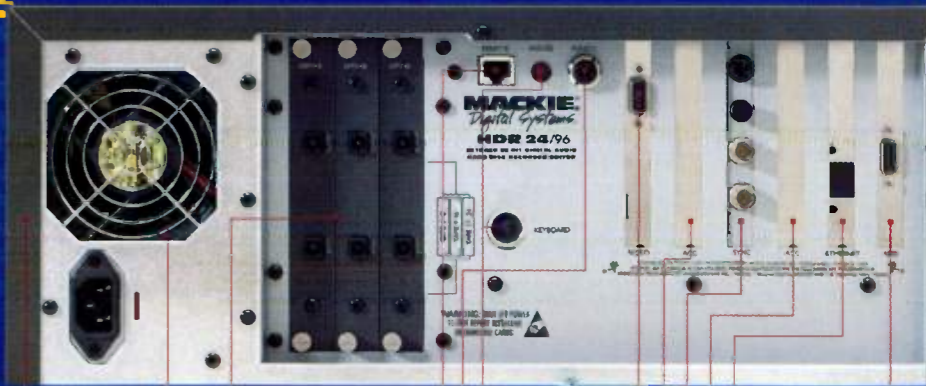
- 4 RU configuration.
- Ultra quiet cooling fan.
- 3-slot card cage for any combination of AIO•8, DIO•8, PDI•8 and new OPT•8 24-bit ADAT optical I/O.
- Remote 24 or Remote 48 PRO input.
- Jack for punch-in/ punch-out foot switch.
- Mouse and keyboard port take standard PC peripherals.
- SVGA video output
- Accessory card slot.
- Sync ports for Word Clock, SMPTE, NTSC and PAL video black burst.
- MIDI port (no extra black box needed).
- 100BaseT Ethernet "HDR Bridge" out.
- Accessory card slot.

HDR24/96 SPECIFICATIONS

Electronic[1]

Frequency response[2]	2 Hz to 22 kHz±0.5dB
Harmonic distortion	0.00001%
Dynamic range	144dB digital / 106dB analog
Adjacent channel crosstalk[3]	−90dBu

Digital

Quantization	24-bit
Sample rates	44.1 kHz / 48 kHz / 96 kHz[4]
Internal CPU	433 MHz Intel® Celeron® w/ 128k internal L2 cache memory
SVGA out	ATI® Rage™ Pro AGP w/8Mb RAM 1024x758 pixels, 72Hz refresh rate
Internal RAM	64 Mb PC66 SDRAM
Ethernet throughput	100kB/sec.

Hard Disk

Internal Capacity	<90 minutes[5]
Mackie M•60 Media Capacity	<90 minutes[5]
Hard disk seek time[6]	9.5 millisecond
Hard disk throughput[6]	295 megabits/sec.

I/O

Mouse	PS/2 style, 6-pin mini-DIN
Keyboard	IBM®-style, 5-pin DIN
Video	SVGA High density 15-pin D-Sub
MIDI	MIDI IN / MIDI OUT, 15-pin D-Sub

I/O cards	OPT•8 ADAT optical DIO•8 ADAT optical & TDIF™ PDI•8 Digital AES/EBU AIO•8 Analog
Nominal / Maximum digital input[7]	−18dBfs / 0dBfs
Nominal / Maximum analog input[8]	+4dBu / +22dBu
AIO•8 A/D Converters	24-bit, 128X oversampling

Physical

Height	7" / 178 mm (4 RU)
Width	19" / 483 mm
Depth	13.25" / 337 mm

Electrical

Operating voltage	120V / 240V / 100 V, 50 /60 Hz
Power consumption	50W

Optional

Remotes	Remote 24, Remote 48 PRO
Future	96 kHz I/O, accessory cards

Irrelevant

Weight of Corporate Chihuahua	3.5lb / 1.6 kg

[1] *with 24-bit digital I/O;* [2] *48 kHz sample rate;* [3] *0dBu at 1 kHz;* [4] *96 kHz via software upgrade and 96 kHz I/O card;* [5] *24 tracks at 48kHz sample rate, your actual time will vary with number of virtual tracks and amount of non-linear editing;* [6] *same for internal and external M•60 media;* [7] *OPT•8, DIO•8 and PDI•8 I/O cards;* [8] *AIO•8 analog I/O card*

Until now, whatever you recorded on a hard disk recorder or workstation was usually stuck inside. Storing, transporting or just changing projects meant a tedious transfer to Jaz® or magneto-optical disk.

With the HDR24/96 you simply pop in a Mackie Media M•60 cartridge and start recording. When you change projects, just change cartridges and put the first one back in its case and back on the shelf (or record to the internal drive and use the pull-out for ultra-fast back-up).

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FIG. 2: This excerpt from a typical *soleares falseta* indicates the way that accents in the *compás* affect the music.

complex, percussive footwork (*zapateado*) and heelwork (*taconeo*) performed with relatively little upper-body movement. Because it demands great physical stamina, this style reinforced a strongly masculine quality. The woman, on the other hand, emphasized graceful, sinuous movements of the hands, hips, and arms. She further expressed her femininity by manipulating her full-length skirt to produce either a sultry or a coquettish effect, depending on the nature of the dance.

These distinctions have remained largely intact except in one important area. Following in the footsteps (so to speak) of dance legend Carmen Amaya, most of today's female flamenco dancers emphasize a great deal more *zapateado* and *taconeo* in their performances. As a result, the exciting sound of complex footwork by both men and women has become an expected part of flamenco performances, and all of the sampling CDs I reviewed include a variety of loops for creating footwork patterns.

Guitar. For a great many musicians, the guitar playing is the most exciting part of a flamenco show. The guitarist (*tocaor*) must provide a compelling and solid accompaniment to the dancers and singers and respond quickly to cues from the performers. The *tocaor* must also know all of the song forms including their rhythmic and harmonic structures. At times throughout a piece (typically at the beginning and during transitions), he provides one or more short solos (*falsetas*)—essentially a series of variations based on the particular metric and harmonic structure of the song form.

Although flamenco guitars resemble classical guitars in several respects, a number of significant features are quite different (see the sidebar "Flamenco Guitar Anatomy"). More important, however, flamenco guitarists use highly idiomatic techniques that are unique to the genre. Unlike classical guitarists (who strive for a clean, elegant style), flamenco guitar players favor an earthy, less polished, more powerful sound that is both bright and dynamic.

Among the most notable flamenco techniques are the various types of strumming (*rasgueados*). These are created by fanning or brushing the fingers across the strings to generate a kind of rolling effect (sometimes quite thunderous) at different tempos and with different rhythms. Rapid, forceful runs producing bursts of notes (*picado*) are another staple of flamenco technique, as is tapping on the top, or soundboard, of the guitar with a right-hand fingernail (*golpe*). This percussive ornament reinforces important rhythms and accents and is frequently used during *falsetas* and accompaniment sections. (Flamenco guitars are equipped with a special plastic shield akin to a pick guard that protects them from *rasgueado* and *golpe* damage.)

Classical and flamenco techniques also differ in their use of the capo (*cejilla*). Although seldom employed in classical playing, the capo is used extensively in flamenco. It allows the guitarist to change keys for different singers while preserving the chord voicings and idiomatic performance elements associated with the various song forms. Because the capo shortens the vibrating string length, it also brightens the guitar's sound; on some pieces, flamenco players may use a capo for that reason alone.

If you're recording your own guitar samples, be sure to include some chords, strums, and arpeggios using a capo at a couple of different frets. Pitch-shifting your samples too much may not provide the desired effect. For *picado* passages, you can use a sequencer to record notes slowly at first, then speed them up later. If you're recording your own samples, however, remember that *picado* notes are always performed with the "rest" or "support" stroke (*apoyando*), as are most notes played with the thumb (*pulgar*).

Speaking of the thumb, flamenco guitarists make much greater use of it than do classical players. In classical music, the thumb is primarily limited to playing the bass notes, while the other fingers concentrate on the treble strings. In flamenco, the thumb ranges over all six strings; in fact, some *falsetas* are performed entirely with the thumb. Treble strings played with the thumb have a slightly heavier tone quality than the same notes played with the fingers. When creating your own samples, you might want to record some separate thumb samples (played *apoyando*) if you're a stickler for detail.

All of the flamenco sampling CDs reviewed here include an assortment of *rasgueados* in different keys, and some also feature *golpe* samples, *picado* notes or passages, and phrases that highlight thumb playing. If you listen carefully, you may hear passages in which groups of notes are played as a single slur
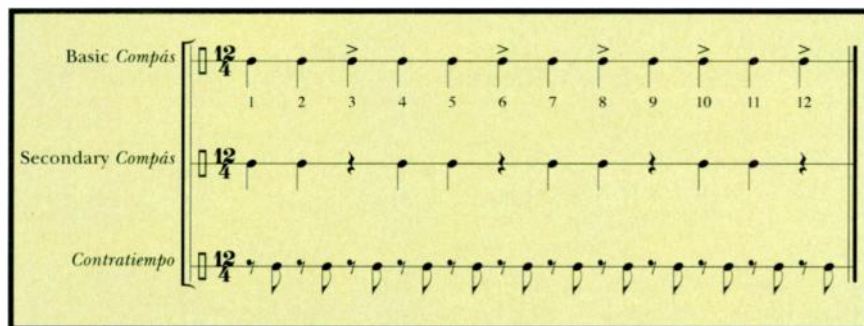

FIG. 3: When multiple players perform *palmas,* they don't always play the same rhythms. In this example, the first player claps the main *compás,* the second player claps a secondary *compás,* and the third player claps the *contratiempo* offbeats.

(ligado). This important left-hand technique is highly characteristic of flamenco playing. So too is the practice of muting the strings with the left hand while strumming with the right. This *palo seco* style of playing transforms the guitar into a versatile percussion instrument for rhythm-only accompaniments.

Percussion. To support the dancers and singers in a flamenco performance, the rhythms are reinforced in a number of ways. The most common and arguably the most important method is through the rhythmic hand-clapping (*palmas*) provided by the other members of the flamenco troupe (*cuadro*). It's important to understand that *palmas* is not simply clapping in time with the music—it's a true art form all its own. It takes study and a great deal of practice to develop the proper

sound (and it takes even more practice to acquire the proper rhythmic sense). If you're thinking of using that "Hand Clap" preset in your General MIDI sound module, forget it. It's not even close to the proper sound.

There are actually two types of *palmas.* The loudest and most forceful is *palmas fuertes,* in which the fingers of one hand (held together) strike the slightly cupped palm of the other hand. When done correctly, *palmas fuertes* produces a loud, clear crack like the sound of a popgun. The other type, called *palmas sordas,* produces a softer, more muffled sound. Here both hands are slightly cupped and the palms are struck together. In general, *palmas sordas* maintains and reinforces the rhythm, while *palmas fuertes* adds an exciting percussive element to the musical texture. When two or more people are performing *palmas,* some of them may begin clapping on the offbeats (*contratiempo*), while others accent different places in the meter. The result is a fast
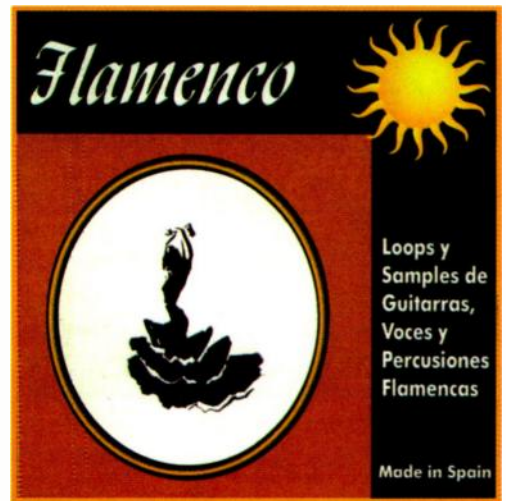
FIG. 4: Discovery Firm's *Flamenco—Made in Spain* offers a good assortment of high-quality samples with an emphasis on *rumba* patterns.

and exhilarating polyrhythmic mélange that may be combined with finger snapping (*pitos*), foot stomping, and other percussive effects. (I'll discuss this topic in further detail a little later.)

When most Americans think of flamenco, they also think of castanets (*castañuelas*). Although they've existed for hundreds of years as part of the Spanish folk-dance tradition, castanets are actually relative newcomers to flamenco. Some purists scorn castanets because they interfere with the beautiful hand movements of the female flamenco dancers. But castanets have become so popular with audiences that they're now widely integrated into modern flamenco performances. Keep in mind, however, that castanets impart a certain lighthearted quality to a piece, so they're most appropriate with folklike song forms such as *sevillanas* and *fandangos de Huelva.* It would be inappropriate and even jarring to insert a castanet part in one of the more profound song forms such as *soleares* and *seguiriyas.*

Unlike orchestral castanets, flamenco castanets come as a matched pair with a slightly fuller-sounding "male" set for one hand and a slightly brighter "female" set for the other. The castanets' different tone colors produce variations in sound as different rhythmic patterns are performed. The dancer may even strike the two sets together to produce an accent.

Castanet rhythms primarily reinforce the prevailing meter and rhythmic structure of the song form. Because of

FLAMENCO GUITAR ANATOMY

Flamenco and classical guitars look very much alike, but some significant differences contribute to their distinctive sounds and allow the various playing techniques employed with each instrument. If you plan to record your own flamenco guitar samples, you'll want to keep these differences in mind. The ideal timbre for a classical guitar is a clean, round, sustained, and elegant sound; conversely, the ideal for a flamenco guitar is strong, bright, penetrating yet earthy. Construction materials as well as structural characteristics are responsible for these and other differences.

To begin with, the body of a traditional flamenco guitar is lighter and not quite as deep as a concert classical guitar. The top is often thinner and the bracing is not as heavy. Whereas rosewood is used for classical guitar bodies, the back and sides of a flamenco guitar are typically made of a light-colored wood (most often Spanish cypress). In recent years, a kind of hybrid guitar has been gaining popularity. These *flamenca negra* models are constructed like traditional flamenco guitars, but with rosewood backs and sides. This imparts a somewhat darker and mellower tone to the instrument while still accommodating flamenco playing techniques.

On a flamenco guitar, the strings are set lower to the fingerboard (the bridge is thinner, and the neck is set at less of an angle), facilitating *ligado* and *rasgueado* techniques and contributing to the instrument's distinctive sound quality. The lower strings may produce some buzzing on the frets during playing, and a little buzz is considered acceptable for a flamenco instrument. In addition, flamenco guitars always have a plastic shield on the soundboard around the string and bridge area. This *golpeador* protects the top of the instrument from damage caused by the energetic finger taps and *rasgueados* that are an important part of flamenco music.

this, and because of the variations in tone color, the castanet presets in most sound modules (whether single notes or rolls) yield disappointing results. It's difficult to create a convincing roll with a single castanet click and a sequencer, and a single-roll sample seldom sounds correct. To solve this problem, all of the flamenco sampling CDs include an assortment of castanet loops at varying tempos and with different rhythms, and some discs come with individual hits so that you can create your own rhythms.

During the past decade, a new percussion instrument has become popular in many flamenco shows. The *cajon* is a large wooden box (measuring roughly 20 by 12 by 12 inches) with a

sound hole in the back and a smooth surface on the front. The player sits on the box and strikes the front panel with his hands. The front panel is mounted in such a way that it produces a slightly raspy or buzzy effect in addition to the wood sound. Striking high on the front panel results in a bright sound; striking lower (toward the center) produces a deep, powerful sound similar to that of a kick drum.

The *cajon* can add another exciting percussive element to a performance, but it also provides important rhythmic reinforcement for the dancers and the other players. It's a testament to the rapid acceptance of the *cajon* by the flamenco community that all three of the flamenco-only sampling CDs I reviewed include an assortment of *cajon* loops for various song forms at multiple tempos.

Now that you're familiar with the vocal and instrumental elements of a
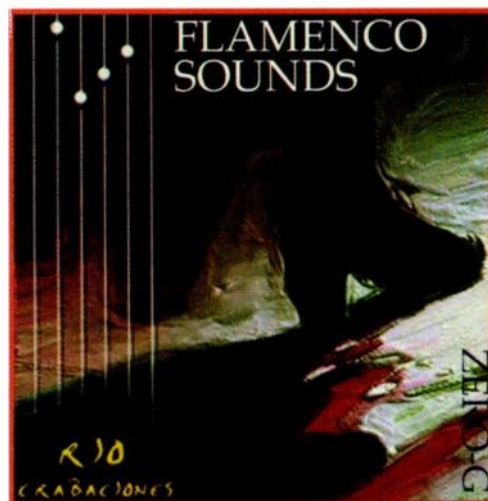
FIG. 5: *Flamenco Sounds* from Zero-G features a broad assortment of musical elements from a number of traditional flamenco song forms.

flamenco performance, let's take a quick look at some of the rhythmic and harmonic elements of flamenco and how they fit together. Afterward, I'll review several sampling CDs to see what they have to offer. (If you forget some of the Spanish terms, see the sidebar "Flamenco from A to Z.")

FLAMENCO FUNDAMENTALS

Although many flamenco songs are sung in a *rubato* (free) style without a definite meter, most guitar- and dance-oriented pieces adhere to a strict rhythmic pattern called the *compás*. The *compás* is a recurring pattern of accents within a set meter that supplies the metrical pulse at the heart of flamenco. The *compás* not only helps define the song form, but also drives the music and galvanizes the performers as well as the audience. Staying "in *compás*" is therefore vital to properly performing most flamenco music and dance.

Simply put, the *compás* is a group of beats with a set of accents superimposed over the group. For example, *soleares* uses a repeating 12-beat pattern with accents on beats 3, 6, 8, 10, and 12 (see **Fig. 1**). The same *compás*—at a somewhat faster tempo—is used in *alegrías*. And at even faster tempos, the same *compás* is used for *bulerías*. Other song forms use other meters: *farruca*, *tientos*, and *rumba* have a 4-beat *compás*, and *malagueñas* and *sevillanas* have a 3-beat *compás*.

Whether you're playing a *falseta* or an accompaniment pattern for dancers, the pulse of the *compás* is always present.

FLAMENCO FROM A TO Z

apoyando	guitar technique also known as *rest stroke* or *support stroke*
baile	dance
cajon	percussion instrument in the form of a large wooden box with a sound hole in the back
cantaor	singer
cante	song
castañuelas	castanets
cejilla	capo
compás	the meter, with accents, upon which many flamenco pieces are based
contratiempo	rhythmic pattern that stresses the offbeats
cuadro	group of flamenco performers
falseta	solo guitar variation
gitano	Spanish Gypsy
golpe	rhythmic taps on the guitar soundboard while playing
jaleo	encouraging words shouted during a performance
juerga	flamenco jam session
ligado	left-hand guitar technique for slurring notes
palmas	rhythmic hand clapping
palo seco	rhythm-only accompaniment
picado	rapid two-finger *apoyando* runs on the guitar
pitos	finger snapping
pulgar	thumb
rasgueado	flamenco guitar-strumming technique
taconeo	rhythmic heel tapping
tocaor	guitarist
toque	flamenco guitar music
zapateado	rhythmic heel and foot tapping and stomping

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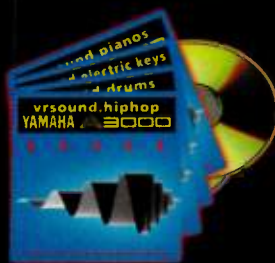
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A single phrase from a typical *soleares falseta* is shown in **Fig. 2**. Notice how the accents affect the melody.

From a melodic and harmonic standpoint, flamenco music is often based on the Phrygian mode, although it uses major and minor scales and progressions as well. Improvising in the Phrygian mode—especially E Phrygian—is an easy way to approximate a "flamenco" sound. The half-step II to I progression is highly characteristic, as is the raising of the third scale degree to make the tonic a major chord, as illustrated in **Fig. 2**.

To get a feel for the *compás,* use your sequencer to step-record 12 quarter notes. Use a *palmas* sample if you have one; otherwise, a hand-clap or similar preset will suffice for now. Set the Velocity of beats 3, 6, 8, 10, and 12 to a value of 127; set the other beats to 50. Now loop the 12-note pattern at a tempo of 80 bpm, and you have a *soleares* metronome. Try improvising against the recurring pattern of accents, and after a while you should get a feel for the underlying pulse. Try playing at different tempos, too.

At faster tempos, other members of the *cuadro* may reinforce the rhythm and increase the excitement level by adding multiple *palmas* parts. In most cases, however, not everyone claps the basic *compás*. In *bulerías,* for example, the basic *compás* is the same as in *soleares* (accents on beats 3, 6, 8, 10, and 12), but there may also be a secondary *compás* with accents on 1, 2, 4, 5, 7, 8, 10, and 11. Meanwhile, other players may be clapping the eighth-note offbeats (the *contratiempo*). The result is a complex overlay of rhythms (see **Fig. 3**).

You can get a feel for how this sounds by using your sequencer. Place the *soleares* metronome that you created earlier into track 1, pan it to the far left, and increase the sequence tempo to 210 bpm. In track 2, enter the secondary *compás* as shown in line 2 of **Fig. 3**. Use a Velocity value of 50 for the notes, and pan this track to the far right. Now take a moment to listen to these two tracks. Can you still hear the

basic *compás?* (Using different sounds for the different tracks may help you focus.) During live performances, the secondary *compás* may be played by stomping a foot instead of by clapping hands.

Next, enter 12 beats in track 3, but displace them by an eighth-note value so that they play on the offbeats. Set the Velocity to 30 for all the notes, and pan the track to the center. Loop the whole thing and practice improvising against it. Experiment with different tempos. Keep in mind that *contratiempo* isn't necessarily used throughout an entire piece—it's added only when the energy level warrants it and is most effective when not overdone. On the other hand, maintaining the basic *compás* through most of a piece with *palmas sordas* or *cajon* is common.

Sequencing flamenco guitar parts from a keyboard can be exasperating (if not impossible) because the music is so "guitaristic." If you have a MIDI-equipped guitar, you can try entering melodies from it. *Rasgueados* are a problem, however. Performing a convincing *rasgueado* on an electric guitar is a real challenge, and what ends up in the sequencer is seldom satisfactory.

As an alternative—especially for keyboard players—check out *Twiddly Bits, volume 3: Electric and Acoustic Guitar* (Mac/Win; $39.95) from Keyfax. Twiddly Bits are short Standard MIDI Files (usually about one measure) that you can combine into longer phrases or use as musical accents. Volume 3 is mainly devoted to folk, blues, rock, and jazz clips, but it also contains 15 MIDI files under the Spanish Strums heading. With a little editing, some of these major and minor strums and *rasgueados* might come in handy, and they don't sound as if they were performed on a keyboard.

For an even more authentic sound, you'll have to use samples of actual flamenco performers. With that in mind, let's take a look at several interesting sample collections.

SPANISH SAMPLES
It may now seem obvious to you, but it nonetheless bears repeating that true flamenco can be created only after years of dedication and hard work.

FIG. 6: Universal Sound Bank's *Flamenco,* a two-disc CD-ROM set for Akai samplers, emphasizes contemporary styles, especially *rumba* and *tanguillo.*

It involves an intimate understanding of the culture's roots and of an art form that encourages spontaneity within complex traditional structures. In other words, the quest for an authentic flamenco sound has no shortcuts. Still, even if you can't fool an aficionado, you can definitely capture some of flamenco's excitement with sampling CDs.

For this article I reviewed four popular titles: Discovery Firm's *Flamenco—Made in Spain* ($99.95), Zero-G's *Flamenco Sounds* ($99.95), Universal Sound Bank's *Flamenco* ($299), and Best Service's *World Colours* ($99.95). The first two are audio CDs that were recorded and produced in Spain. The third, recorded in Spain and France, is a two-disc CD-ROM set for Akai samplers. The last title is a two-disc audio-CD set produced in Germany that includes a flamenco section; it's also available in Akai format (one disc, $199.95). All of the titles were recorded with experienced Spanish performers who impart the proper styles and techniques to the phrases, loops, and individual samples.

Flamenco—Made in Spain. Recorded in Madrid, Discovery Firm's *Flamenco—Made in Spain* (see **Fig. 4**) offers a good assortment of guitar, vocal, and percussion samples and loops. The recording quality is generally quite good; most of the samples are close-miked and clean. The lack of heavy reverb gives the recordings an intimate quality and allows you to add your own processing later.

A quick look through the contents

Roast, Toast, Grin, and Boast

The CD-R Pro and CD-R Deluxe, TASCAM's third generation CD Burner Bundles, offer the best combination of powerful software, ease of use and advanced features. Combining Macintosh and Windows software with a dual platform SCSI card in the same box, both bundles offer the ability to record audio and data CDs and provide 24-bit/96kHz capabilities with advanced real-time processing. Add to that award winning TEAC CD-R technology and you've got the most flexible, powerful, do it all CD-R bundles available at an affordable price.

The Pro bundle is designed for the user who wants it all. Complete with a TEAC 8x24 CD-R drive, this bundle features a collection of software to record audio and edit tracks. With an extensive list of PQ code controls and 100% Red Book compliant masters, this software suite also provides advanced real-time audio processing. For creating MP3s, Xing Audio Catalyst is included for both platforms offering the best sonic quality available and 100% compatibility with most MP3 players.

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CD-R Pro and Deluxe Burner Bundles

reveals that this collection is at least partially slanted toward a pop/Latin/ flamenco fusion style of music with lots of *rumba* patterns and even bongo samples. This CD should prove to be a great resource for creating music in the style of the Gipsy Kings. Plenty of traditional flamenco phrases and loops are featured throughout the disc as well.

The collection begins with several *rasgueados* in different keys followed by several contemporary-sounding *rumba* comping patterns with two guitars. The next group of samples includes an excellent full-length *bulerías* section (consisting of several *falsetas*) followed by some very fine *bulerías, alegrías, seguiriyas,* and *soleares* phrases. The guitar work of Pepe Nuñez successfully conveys a true flamenco style and sound. After these phrases come a variety of two-guitar *sevillanas* fragments that many people will find useful for creating songs and upbeat styles of music.

The next large section features an extensive collection of male, female, and group vocal samples, including sung phrases, spoken expressions, and some wonderful *jaleo*. Many of these samples are suitable for looping; others could work well sprinkled throughout a piece for added excitement or humor. Percussion instruments include standard bongos, Moroccan bongos, and *cajon*. All are played in a variety of *rumba* patterns and with individual hits and flams. The bongos are of questionable value for traditional flamenco but may be useful for flamenco-fusion pieces. The individual *cajon* hits will be especially handy if you wish to perform your own *cajon* parts for a flamenco composition.

The disc has a generous assortment of solo and group *palmas* loops, many with *contratiempo* rhythms. Single claps are provided, too. In addition, you get finger snaps, castanets, and male footwork in numerous loops, phrases, and single hits. I particularly like the extended *zapateado* solo.

Guitar playing is well represented in the collection, with dozens of *picado*

phrases (including some thumb work), accompaniment patterns, and *falseta* fragments. It's too bad that the *falseta* passages aren't identified by their song form—that might make them easier to use. Finally, this disc is unique in providing a number of muted *(palo seco)* guitar patterns as well as muted flams and single hits.

Overall, *Flamenco—Made in Spain* offers an excellent assortment of flamenco samples that could be combined in many interesting ways. My only gripe is that the accompanying booklet lacks key and tempo indications for many of the samples. More descriptive information would help a great deal. Nevertheless, this collection has much to recommend it, especially for fans of *rumba*.

Flamenco Sounds. Unlike *Flamenco—Made in Spain*, Zero-G's *Flamenco Sounds* (see **Fig. 5**) does not emphasize *rumba* rhythms and loops. Instead, this CD offers a broader assortment of musical elements from several traditional flamenco song forms. Although the recording quality is not as clean or consistent as that of *Made in Spain*, many performances are quite good. Moreover, nearly all of the entries are labeled with a key indication, as well as a tempo marking where appropriate.

The CD begins with 36 short guitar phrases from a variety of song forms including *alegrías, bulerías, fandango, soleares,* and *tango*. Many of the phrases were derived from traditional *falsetas* and have a highly characteristic sound. The next section offers more than two dozen arpeggio patterns, in addition to rhythms with arpeggios in different keys and tempos. These phrases are well played and appealing and could be used in a number of settings. The small group of tremolo phrases, however, is a bit disappointing—the phrases are performed too disjointedly to provide the smooth, continuous sound needed for a good tremolo effect.

Flamenco Sounds includes more than 36 phrases and fragments emphasizing thumb work, and several include *ligado* passages. Many of the entries are quite good and provide a real flamenco sound. The same is true of the 51 *picado* phrases. These aren't particularly loop-

FIG. 7: Best Service's *World Colours* offers a variety of sounds from around the globe, including a section of flamenco samples.

able, but you can easily combine them in different ways using a building-block approach. The remaining phrases in the guitar section combine several techniques including *picado, rasgueado, ligado,* thumb variations, arpeggios, and *palo seco* rhythms.

One of the CD's biggest strengths is its male singing section, which offers phrases from a number of traditional flamenco songs. The singing of Rafael Silva "Chocolate" provides the proper pathos with an evocative and earthy quality. The female vocal section is not quite as impressive: the singer's voice is too velvety and languid to generate enough excitement or emotion in many of the phrases. A section of *jaleo* shouts and expressions, though, is quite good and should provide plenty of colorful material for your next flamenco piece.

Another of this collection's strengths is the *cajon* section. The recording quality is better here than in some of the other sections, and the loops and phrases are excellent, especially the fast *bulerías* patterns. This section also includes a variety of hits, flams, and other short fragments.

A section of group *zapateado* offers some fine dancing (sometimes with *palmas*), but the recording includes a great deal of reverb—it sounds as if it were recorded in a dance studio. Consequently, blending these phrases with other tracks is difficult. Solo *zapateado* phrases are also provided, and the performances are excellent, although

Desktop Flamenco

some of the phrases are too quiet.

Flamenco Sounds has a terrific *palmas* section with several exciting phrases that include *contratiempo* and foot stomps. The castanet section offers nearly two dozen very nice rhythmic patterns, but all of the samples include *palmas sordas* as well, so you can't get a clear castanet phrase if you need one—very unfortunate.

Although this collection is uneven in places, it boasts some strong sections with several excellent samples. Moreover, its creators have made an effort to represent a variety of traditional flamenco song and dance forms, adding to the disc's value.

Flamenco. As with *Flamenco—Made in Spain,* Universal Sound Bank's *Flamenco,* a two-disc CD-ROM set for Akai samplers, emphasizes contemporary styles, especially *rumba* and *tanguillo* (a close relative of *tango*). Unlike the other discs, however, *Flamenco* (see **Fig. 6**) offers little in the way of traditional song forms such as *bulerías, alegrías,* and *soleares.*

The first disc is devoted entirely to guitar strumming patterns, which are offered in major and minor keys at four tempos: 100, 112, 119, and 133.5 bpm. In addition, each pattern includes a *palo seco* version at each of the tempos. The disc also provides a few solo *rasgueados* and some handy ending patterns. The recording quality is good and the comping patterns loop nicely (although they don't immediately suggest a strong flamenco feel). These rhythm guitar loops might work well for folk, pop, Latin, or mariachi music—and of course, they could be used for flamenco-style *rumbas.*

The second disc begins with several arpeggios in major and minor keys. The patterns loop well and sound great, but the performances sound a bit too "classical"—they lack that flamenco edge. The disc also includes individual notes with Velocity-switching between hard and soft attacks. Other

Improvisation and spontaneity are important elements in flamenco.

banks offer chords strummed slowly up or down, guitar slurs (with Velocity-switching), and solo phrases at different tempos for *rumba* and *tanguillo.*

The *cajon* section is one of this collection's finest features. It includes a wide variety of loops at the same four tempos as the guitar accompaniments—mostly for *rumba* and *tanguillo.* It also provides a good assortment of single hits, rolls, flams, and other effects for creating your own rhythms. Other banks contain some very good *zapateado* loops and single hits, as well as several excellent castanet loops (at the same four tempos) and single hits. I especially like several of the *palmas* presets, which include loops and short samples that you can easily combine into *contratiempo* rhythms once you get the hang of it.

A section of male and female singing offers a variety of short phrases for *rumba* and *tanguillo* in the established four tempos. Again, the male singing is a bit stronger than the female, but both are somewhat limited by the disc set's exclusion of other song forms. Unlike competing sample CDs, *Flamenco* comes with no lyrics—not even the opening lyrics—for the phrases, making them much harder to find and use. Other vocalizations include a section of male and female *jaleo* and nine samples of the *cuadro* milling around while talking and kidding with one another in Spanish.

This collection is rather limited in terms of the song forms that it incorporates, but it does include a number of excellent samples. Furthermore, because most of the samples conform to the same four tempos, mixing and matching loops should be relatively

FLAMENCO ON THE WEB

Fernández Music
www.fernandezmusic.com
Ron Fernández is a wholesaler of guitars and flamenco accessories. His site features numerous informative articles on flamenco-related topics.

Flamenco Central
flamencocentral.homepage.com
This Southern California–based site features information and links to flamenco teachers, performers, and local events. It also offers pictures of flamenco performers and contacts in Spain.

Flamenco Connection
flamenco.bizland.com
This site provides contact information for Flamenco Connection, a mail-order retailer in Virginia that specializes in flamenco CDs, books, videos, instruments, strings, and accessories. Flamenco Connection (888-FLAMENCO) has an extensive and impressive inventory, and it's well worth your time to contact the company and request a catalog.

The Flamenco Guitar Homepage
www.guitarist.com/fg/fg.htm
This site provides information for aspiring flamenco guitarists and also offers printed excerpts of music and links to other sites.

Flamenco World
www.flamenco-world.com
This important site, based in Spain, has a wealth of information about flamenco and its performers. In addition, it features RealAudio clips and links to sites that offer CDs, books, and flamenco accessories.

The German Flamenco Pages
www.multimania.com/germanflamenco/contents.htm
This site has a ton of information on flamenco, including news, links, music, contacts, and much more.

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easy. But if you're looking for a broad range of traditional song forms, *Flamenco* may disappoint you, despite its otherwise fine samples.

World Colours. As the name suggests, Best Service's *World Colours* (see **Fig. 7**) offers a variety of samples from around the globe. In fact, this two-CD set is jam-packed with a surprisingly wide range of musical excerpts and fragments from such faraway locales as Russia, Bulgaria, Greece, Turkey, India, Bali, and Africa. The Gypsy *(Romani)* selections from Eastern Europe include accordion, violin, and zimbalon samples as well as wind instruments, vocals, and percussion.

A modest flamenco section with an assortment of guitar strums, grooves, and solos begins the first disc. Some of the samples are so short (little more than a fraction of a second) that they're of questionable value. However, some of the longer patterns might make nice accompaniment parts. Most of the samples sound as if they were extracted from a recording of a larger performance: several clearly have *palmas* and other sounds in the background—that could cause trouble when looping and building tracks. In general, the guitar samples are presented as a potpourri of styles and sounds, with *rasgueado, picado,* and other techniques offered in no particular order.

Rhythm patterns using the guitar body as a percussion instrument (often heard during flamenco performances) are provided here, but the collection lacks the muted guitar sounds common in *palo seco* passages. A small section of group *jaleo* is offered, as are several short phrases sung by male soloists. The singers create a good *gitano* sound, but in several of the phrases guitar, *palmas,* and other elements are audible in the background, limiting the samples' usefulness. Furthermore, the recording quality ranges all over the place: some phrases sound relatively clean and close, while others sound more like field recordings. The same problem plagues many of the guitar samples.

A section of rattles and shakers is of limited use for flamenco (a *cajon* section would have been a welcome addition), but a number of *palmas* and *zapateado* samples have creative potential. Several samples combine *palmas* with vocals, *jaleo,* and footsteps, so they aren't particularly suitable for looping. Some of the extended *palmas* samples are quite good, with hand claps and foot stomps generating characteristic *contratiempo* rhythms. *World Colours* also offers individual *palmas* hits, but most were recorded with too much reverb. Group and solo *zapateado* patterns are included, and some are quite good, but they suffer from the same recording-quality inconsistencies as the earlier samples.

In general, the flamenco section of *World Colours* is a hit-or-miss grab bag of samples. The collection's value is unfortunately diminished by the poor organization of the discs, not to mention the lack of printed information: the flamenco section offers no tempo markings, key indications, lyrics, or identification of song forms, and the performers aren't even credited. Still, *World Colours* has a few gems, including some excellent *palmas* with *contratiempo,* some very good extended *zapateado* patterns, and a few nice guitar phrases.

LOOK AND LISTEN

As you may have guessed, the art of flamenco encompasses far more than can be covered in a single article. If you aren't already familiar with the music, buy a few flamenco recordings and acquaint yourself with it. Most large CD retailers have a section for the music of Spain; you should have no trouble finding plenty of recordings from the past and present.

Many books and instructional videos are also available and worth exploring. If you don't know where to look for flamenco resources, start with your Web browser. The Internet is packed with flamenco information, and you can order anything from CDs to castanets online or over the phone. (For a list of places to start, see the sidebar "Flamenco on the Web.") Finally, keep an eye open for flamenco performances in your area. Watching and listening to live flamenco is unquestionably the best way to capture the spirit of this exciting art form.

EM associate editor **David Rubin** *lives and works in the Los Angeles area, where he is currently studying flamenco guitar with Paco Arroyo.*

Cartoon Music Editing for TV Animation cutups:

By Erik Hawkins

The wacky undersea world of *SpongeBob SquarePants,* Nickelodeon's first original Saturday-morning cartoon, is officially a hit. Recently touted by *TV Guide* as the most watched Saturday-morning kids' show, this highly original series springs from the mind of marine biologist Steve Hillenburg—which explains the show's oceanic slant. It features a host of kooky saltwater characters: the protagonist, SpongeBob, a pants-wearing, rectangular, bucktoothed sea sponge with goo-goo eyes; his pet snail, Gary; his best friend, Patrick the starfish; an egotistical clainet-playing neighbor, Squidward; and SpongeBob's love interest, the beautiful Sandy Cheeks, a squirrel who lives in an underwater biodome. The show's theme is simple: SpongeBob's simpleminded antics, although well intentioned, continually wreak havoc. ● With such a unique cast and setting, the show has ample room for an equally unusual musical score. But Nickelodeon isn't hiring famous composers to perform original music for every episode. *SpongeBob SquarePants* is a new show wading into untested waters (no pun intended), so budget constraints are a real consideration. And the madcap production pace of an animated series—cutting all the music for a 30-minute episode usually takes about a week—leaves little time for writing original music.

● Enter Nick Carr, an ace music editor able to weave mind-blowing musical backdrops with only a small collection of home studio gear and some *needle-drop music* (prerecorded music purchased for commercial use, also known as *production music*). During his two decades in the business, Carr has received accolades including an Emmy and a Golden Reel Award (given to outstanding editors by the Motion Picture Sound Editors). He has lent his talent to shows such as *The Really Mighty Ducks* (Disney), *Pepper Ann* (Disney), *Rocko's Modern Life* (Nickelodeon), *Modern Marvels* (Discovery Channel), and *Power Rangers* (Saban Entertainment). Carr's editing expertise and creative use of music in *SpongeBob SquarePants* have garnered much applause, most recently a 1999 Golden Reel Award for music editing in an animated television show.

Top music editor Nick Carr works his magic on Nickelodeon's hit cartoon *SpongeBob SquarePants*.

Cutting music in a personal studio for a top-rated children's cartoon.

Cartoon cutups

CORNER STUDIO

Upon first hearing the *SpongeBob* soundtrack, most people would never imagine Carr's studio to be the humble space that it is. Housed in a small abode precipitously perched in Los Angeles's Studio City hills (where he lives with his wife and business partner, Chiho, and their wiener dog, Oscar), Carr's entire studio fits into one corner of his living room. In the opposite corner, taking up almost as much space as his equipment, are two large bookshelves packed with music-library CDs. Within these four walls, Carr does some of his best work—a stone's throw away from multimillion-dollar movie and television studios such as Disney, Warner Brothers, NBC, Nickelodeon, and Universal Pictures.

Carr's system centers on a SoftSplice 4-track hard disk editing system hooked up to a Macintosh Quadra 650. The SoftSplice is a little-known piece of gear from the now bankrupt company Digital Expressions. A 19-by-5.25-inch rack unit, the SoftSplice communicates with the Mac via SCSI. It handles all of the audio files and processing; the computer functions solely as a front end. Proprietary software installed on the Mac controls all of the SoftSplice's functions (see **Fig. 1**).

The SoftSplice itself has only digital I/O (AES/EBU and S/PDIF); it has no
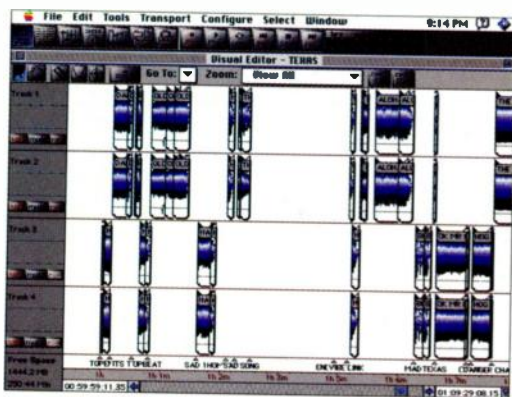
converters. The unit uses a companion 18-bit A/D/A converter for monitoring, and a Tascam DA-30mkII DAT recorder acts as both source and backup medium. (You can back up the SoftSplice's internal hard drive to DAT.) The converter is normaled to the AES/EBU input and the DAT to the S/PDIF; you can select I/O formats directly from the interface software. The machine can read and write SMPTE in a variety of formats (24, 25, 29.97, 30, and 30 drop).

A ¾-inch Sony VP-9000 video deck (for playback only) that Carr bought used for $600 feeds a 32-inch Toshiba television (which doubles as the household TV set) to provide picture. Professionals in film and television post-production generally use ¾-inch video work tapes, although ½-inch VHS tapes are also common, and digital video is quickly gaining popularity.

The work tapes are striped with 29.97-frames-per-second (fps) non-drop-frame SMPTE time code. The SMPTE track acts as the studio's master sync source: a Y-cable splits the signal and routes it to both the SoftSplice and a Tascam DA-88 digital multitrack recorder outfitted with an SY88 SMPTE card. All music is delivered on DA-88 tapes, an industry standard.

MIDI sound modules, samplers, and effects units fill a waist-high rack shelf (see the sidebar "Modest Digs" for a complete list of Carr's gear). Everything runs through a 24-input Roland M-24E console and is monitored on a pair of Boss MA12C self-powered speakers. About 10 inches tall, these diminutive speakers are only a little bit bigger than typical multimedia speakers. Carr says that the motion picture industry uses these widely as a common point of reference, and he stresses that since he isn't actually mixing the music, high-end monitors aren't necessary. The MA12Cs also emulate the sound of TV speakers, through which most people hear *SpongeBob*.
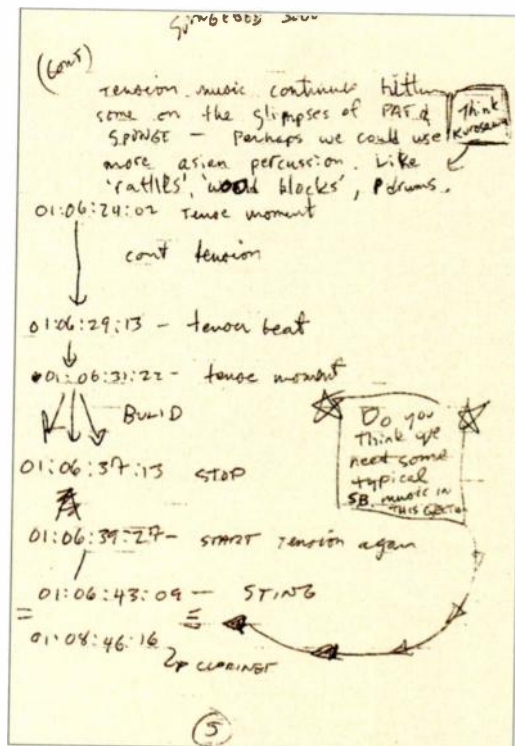
FIG. 2: The spotting notes tell Carr what the director wants in terms of the music, moods, genres, effects, I/O times, and other considerations.

FIG. 1: Proprietary software allows the Macintosh to act as the front end of Digital Expressions' SoftSplice 4-track digital editor, controlling all transport, editing, and DSP functions. The SoftSplice connects to the Mac via SCSI.

COUNTDOWN TO PICTURE

Two 11-minute segments compose a 30-minute show; the other 8 minutes are dedicated to commercials. For Carr, the editing process begins when the director, Steve Hillenburg, faxes the *spotting notes* to him (see **Fig. 2**). (Spotting is the process of viewing picture and identifying the parts that need sound or music.) These notes detail what type of music each cue should be and where the cues should appear. Shortly after the fax arrives, a courier delivers a ¾-inch work tape of the final picture and rough dialogue.

In animation, the dialogue is recorded first, and then the visual element is created to match it. Sometimes the rough dialogue is kept for the final mix; other times it's rerecorded in a process called *automatic dialogue replacement* (ADR). Once the dialogue is done, Carr begins to lay in the music for the final product. At the same time, Jeff Hutchins, the series' sound effects designer and editor, is busy laying in the appropriate pops, squeaks, explosions, and footfalls. Rarely does Carr hear the sound effects before the final mix.

At this stage, Carr cuts the first round of music according to Hillenburg's

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notes. This is often easier said than done, for although the spotting notes are nicely detailed with SMPTE I/O times, the timing is usually off. By the time Carr gets the work tape, Hillenburg's spotted times have shifted, often due to picture edits. If Hillenburg cuts out several frames in the middle of the show, the rest of the picture moves up by that number of frames, and nothing beyond that point corresponds to the director's notes. Carr has no magic solution to this problem. "I just have to pay close attention to the description in the spotting notes and trust my intuition."

MODEST DIGS

Carr's modest but effective setup includes the following equipment:

Computer Gear
Macintosh Quadra 650
Opcode Studio 3 MIDI interface
Mark of the Unicorn *Performer*
Digital Expressions SoftSplice DAW
JLCooper PPS2 sync box

Video Gear
Toshiba 32-inch television
Sony VP-9000 ¾-inch U-Matic

Recording Gear
Roland M-24E mixer
Tascam DA-88 MDM
Tascam DA-30 mkII DAT
Boss MA12C powered monitors

MIDI Controllers/Processors
Roland Octopad 8
Oberheim Strummer

Samplers and Sound Modules
E-mu ESI-4000
E-mu Emulator II
Kurzweil MicroPiano
Korg M1r
Oberheim Matrix 6R
E-mu Pro/Cussion

Signal Processing
Yamaha Rev7
Lexicon PCM 70
Alesis MicroVerb II
BBE 262 Sonic Maximizer

CHOICE CUTS

Interpreting the director's notes and choosing the music he envisions is always a challenge. Because the show takes place underwater, Carr uses a lot of surf music for action sequences; "'Wipe Out'–type music," he says. For dreamy situations and everyday walk-and-talk scenes, Carr picks straight-ahead Hawaiian tracks. With dramatic moments, he opts for music that's "way over the top, like a London Symphony Orchestra track," and to convey a sense of heavy danger, he pumps up the volume as much as possible with string swells, *piatti* (cymbals), timpani rolls, and so on.

SpongeBob's producers have completed 20 half-hour segments. The process has gone smoothly so far, and Carr is quickly amassing a library of music that Hillenburg likes. Carr says that this speeds up cue choices. "Knowing what's going to work and what isn't gets a little bit easier the longer you work on a show." But the show's musical needs are always changing. Carr explains, "Though we have a set of staple music we regularly pull from, we are constantly introducing new musical styles as the characters evolve." For example, recent episodes have incorporated country music, an electronica dance piece, and a track commissioned from a local independent band.

Sometimes the director's notes are quite explicit. For instance, the cue for a scene in which SpongeBob is sleeping reads, "Nighttime Hawaiian music." Carr recalls, "I've used a Hawaiian steel track for this type of description before, and the director loved it." The cue is only about 10 seconds long, but the CD track is almost 5 minutes, so the track is perfect for recycling. Carr can cull from it quite a few 10-second segments. "No matter how I cut it, it always comes out different," he remarks.

Occasionally cue descriptions are vague at best. One reads, "Title cut, sickness," followed by a big question mark. (The title cut is the first 10 seconds of the cartoon, the intro and title before the story.) "This is a tough one," says Carr. "I haven't done it yet."

Music editors often leave title-cut music for last, because it sets the tone of the show. After all of the other music has been selected, the episode has a

SpongeBob SquarePants, with its wacky characters and innovative story, requires equally creative music.

set musical feel to which the editor will match the title cut. Carr looks for a signature piece representing the episode's subject. He might spend an entire day combing through music libraries to find the perfect track. But the director may hate Carr's first choice, so he lays down several different cues.

Periodically Carr needs to play a piece of music himself. In one episode, for instance, SpongeBob sings to a hypothetical beat without any backing track. Carr needed a piece in rhythm with SpongeBob's vocal riffs. Finding prerecorded music that played in time with SpongeBob's actions would have been nearly impossible, and time-stretching or cutting audio to match picture is a real pain.

At times like this, Carr's experience as a drummer comes in handy. He pulled up a conga sound on his E-mu Pro/Cussion drum module and, with the SoftSplice locked to picture, played the part on a Roland Octopad drum controller, recording the result directly to disk. Carr recalls, "After a few tries I nailed it, and voilà—the cue was done." Carr works faster live and does so whenever possible, but he occasionally uses Mark of the Unicorn's *Performer* sequencer.

TRICKS OF THE TRADE

Carr often employs pitch-shifting and samplers to make keys and tempos

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ECHO

LAYLA 24/96

ECHO

GINA 24/96

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ECHO

DARLA 24/96

LINE INPUTS   LINE OUTPUTS

ECHO

MONA 24
DIGITAL STUD

echo
digital audio

Cartoon cutups
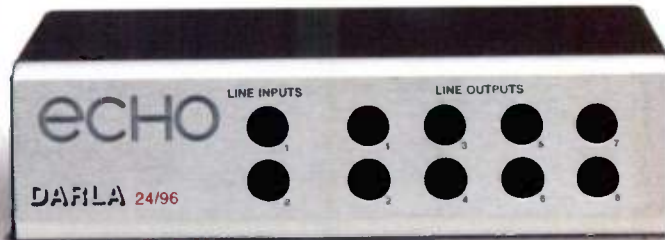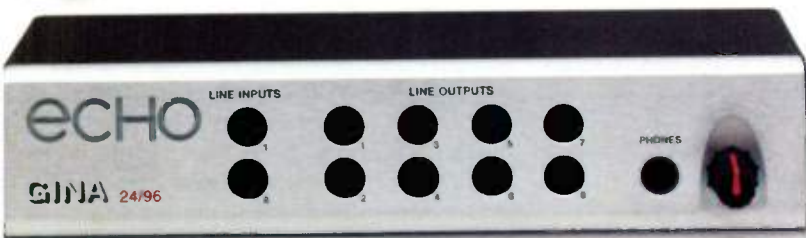
match. "Since this is a toon, I can get away with a lot, speeding things up and slowing them down." Carr sometimes gradually speeds up a cue as onscreen action gets crazier. He says, "This technique works great, thanks to the Soft-Splice, which has a real-time speed control that sounds like an analog deck." Carr simply slides the Soft-Splice's onscreen Speed fader forward or backward to change a track's tempo and pitch (see **Fig. 3**). Sometimes he does this while viewing picture and recording directly to the DA-88; at other times he records the effect to DAT and then flies it back into the Soft-Splice for editing.

A more radical effect than the Soft-Splice's Speed control involves tweaking a track with a sampler's pitch and modulation wheels. Carr selects a piece of music, samples it into his E-mu ESI-4000 sampler, then moves the pitch and mod wheels while the sample is playing. After achieving a sufficiently warped and warbled sound, he records it to DAT. Because the DAT recorder connects digitally to the SoftSplice, that's a breeze.

Over the years, Carr has collected extensive Emulator II and ESI-4000 sample libraries, which he often draws on for orchestra hits and sampled loops. He uses these samples as links between cues. If a cue ends seconds before another one starts, Carr may trigger a sample of a timpani roll or cymbal hit

to bridge the gap. In this way, he can connect cues with different tempos and keys. Carr might also manually play a string or other instrumental part so that it speeds up or slows down to match the upcoming tempo, simultaneously ascending or descending in pitch to match the keys.

Carr runs the tracks through outboard gear on a case-by-case basis; some cues need weird effects, some don't. If he comes across a poorly recorded sample or piece of music, Carr runs the track through his BBE 262 Sonic Maximizer to give it more clarity and crispness. He adjusts delays and reverbs in real time as he pipes music through the effects. He usually records the processed track directly to DAT, later porting it to the SoftSplice for placement.

Cool effects can come from the most unlikely sources. Recently the Soft-Splice drive crashed in the middle of a show and began playing back the music tracks in a crazed, fragmented fashion. The bizarre audio glitch was the perfect musical effect for a time-machine sequence. Carr sampled the sound, doctored it up with some delays and additional music beds, and flew it into the show. "The director nearly fell out of his chair—he loved it."

DIRECTOR'S CHOICE
After Carr completes the first round of music edits, Hillenburg swings by his pad and reviews the work. Hillenburg watches the episode straight through, silently taking notes. Then it's time for Carr to press the rewind button and address the director's changes. "At that point I'm doing really quick fixes—fast, rough edits—because the director doesn't have time to sit around while I fine-tune. That happens after he's left," Carr says. If Hillenburg wants a totally different sound, Carr
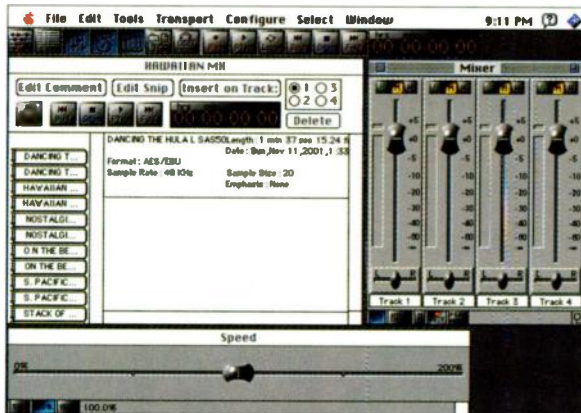
FIG. 3: SoftSplice's audio-editing software includes a Mixer window and a handy Speed control.

pulls out piles of CDs and runs through alternatives. Instead of loading each cut into the editor for playback, he cues it up on the CD player. "I wild-sync it to picture," he says, "just to get a feel for whether the track will work." If Hillenburg likes the cut, Carr makes a note of it and they move on to the next cue.

When Hillenburg hits the road, Carr loads in the new cuts and adjusts the necessary cues; with all elements still in the editor, making changes is a snap. Afterward he lays all the edited music tracks back to the DA-88. He alternates successive cues between two stereo pairs of tracks, a process known as *checkerboarding*. For example, if the first cue records to tracks 1 and 2, the second will record to tracks 3 and 4, the third will record to tracks 1 and 2, and so on. **Fig. 1** shows how the music cues appear on alternating tracks in the Soft-Splice before going to the DA-88. The SoftSplice provides only four tracks, so Carr sometimes has to make two passes to the DA-88. "But since it's just a toon, not a feature film," he says, "four tracks is usually enough."

Carr must use more than four tracks when the director isn't sure about a cut or has chosen one different from Carr's original selection. In such a case, Carr prints other options on additional tracks. He keeps his first choices on tracks 1 and 2 and prints other selections on tracks 5 through 8. Each cue has a nearly identical SMPTE start time, allowing Hillenburg to make a decision in the final mix. "There have been times," says Carr, "when the director didn't like my selection and chose another bed during our review, but in the mix he decided my cut actually sounded better." By keeping the alternate takes

Nick Carr (left) in the studio with sound effects editor Jeff Hutchins (center) and director Steve Hillenburg.

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WRH

Cartoon cutups

available on other tracks, Carr covers all the bases. Says Carr, "It's better to have too much than not enough."

DUBBING STAGE

After Carr has tracked everything to the DA-88, he sends the tape to the dubbing stage for mixdown. The dubbing stage is a full-fledged recording facility with a 48-input automated mixing board, racks of expensive outboard gear, an exotic microphone collection, comprehensive digital-editing capabilities, a coffee-break room, lots of Emmys on the walls—you get the picture.

When Carr's music is locked to the sound effects and final dialogue, and these elements are locked to picture, the show is ready for mixing. The two mix phases are *premix* and *final mix*. The premix, handled by a second engineer (usually after-hours), provides a base mix from which the main mix engineer, Tim Borques, can create the final mix. (Here the term *premix* means a rough version of the final mix, but the film world also uses *premix*, along

with *predub*, to indicate the process of mixing a large collection of edited tracks down to a more manageable number.) The director and his assistants are present when Borques tweaks the final mix. Carr usually stops in toward the end of the process to check out the result.

Carr says, "A lot of credit goes to Tim, because he ends up with everyone's material: dialogue, effects, music, everything—he's the one who puts it all into perspective." To help Borques decipher what's going on, Carr provides a cue sheet displaying the exact times and titles of all the cues. In the margins, he writes notes about special treatments; for example, "Make this cue sound like it's through a telephone handset," "The music comes from a passerby carrying a boom box," and "This is heard on a TV in the living room." Carr leaves the cues dry, usually without fades, and provides some overlap time between cues on the alternating tracks. This gives the mixer plenty to work with when crossfading or adding effects and reverb tails.

THE FINAL TASK

By now Carr has spent countless hours choosing and editing music for the episode, but his job still isn't over when he hands the tape to the mixer. In order for the composers and copyright-holders of the music libraries to receive

Project:	SpongeBob SquarePants			MUSIC CUE SHEETS			
Production #:	Pilot			Prepared by:			
Title:	Help Wanted			June 9, 1999			
Producer:	Nickelodeon						

Cue #	Cue Title	Time	Usage	Composer	%	Publisher	Perf. Rights
1	KPM 155-69	0:02:00	I/b			APM	
2	KPM 155-69	0:04	I/b			APM	
3	CD 77089-01	0:12	I/b			APM	
4	CD 77089-01	0:29	I/b			APM	
5	CD 77089-07	0:13	I/b			APM	
6	CD 77089-07	0:05	I/b			APM	
7	SCD 221-05	0:06	I/b			APM	
8	SCD 221-48.2	0:01	I/b			APM	
9	SCD 221-48.2	0:06	I/b			APM	
10	SCD 221-48.2	0:03	I/b			APM	
11	SCD 221-01	0:14	I/b			APM	
12	CD 77089-07	0:06	I/b			APM	
13	CD 77089-07	0:07	I/b			APM	
14	CAR 176-64	0:16	I/b			APM	

FIG. 4: The publishing cue sheet lists all of the final times and cuts used for a show. Meticulous accuracy and detail are extremely important in this cue sheet, because it determines the payment of publishers and composers.

residuals, Carr must meticulously fill out a publishing cue sheet (see **Fig. 4**).

The publishing cue sheet lists the final tracks' times, titles, and composers, as well as the music libraries from which the tracks were culled. Carr also notes the music's use: whether it's an instrumental background (for example, backing music for people dancing) or a featured instrumental (for instance, music with sung vocals). Residuals are much greater for featured instrumentals than for background music. If Carr lists the wrong composer, incorrectly describes his use of the music, or makes a mistake with any other item on the publishing cue sheet, a composer may not get paid the amount that he or she deserves. Carr says, "I always try to be as clear and concise as possible. People's residuals are on the line."

Once the publishing cue sheet is complete, Nickelodeon's legal department fills in some additional details, such as whether a publisher is with BMI or ASCAP. Music editors don't always need to fill out the publishing cue sheet, but for *SpongeBob*, says Carr, "That's the way it works. It's bothersome and time-consuming, but essential." In the end, Carr is the only person who knows what all the music cuts are and where they came from.

Carr's favorite music library for *SpongeBob* is Associated Production Music (APM). Nickelodeon purchased the rights to use APM, giving Carr open access to all of its tracks. Although other libraries are available, APM has perfect toon tracks, says Carr, "stupid, corny, over-the-top stuff." He mentions that many of the libraries he's heard are "too clean, too perfect, too pasteurized." For *SpongeBob*, Carr needs a lot of live vintage stuff from the '60s and '70s, and APM has this type of material. Whenever possible, the director also makes songs from independent artists available by buying the rights to use their tracks or albums, such as those of Los Mel Tones and the Woodies (both distributed by LoveCat Music in New York). The director envisions using more tracks from independent artists as the show progresses.

CUT AND FADE

Carr offers this advice to would-be music editors: "Get your rhythms down. Take some drum lessons, rhythm guitar—whatever it takes—and learn your time signatures and key signatures." It's not just about grabbing a piece of music and pasting it in, he says. "It's about tempo and feel, hitting the right spot in the right key to make completely different types of music, recorded in completely different locations, blend seamlessly together."

Carr's accomplishments prove that success comes not from having the most expensive, newest gear, but from getting the most from what you have. A ¾-inch deck, a Tascam DA-series recorder, a DAT machine, and an editing system (with at least four tracks and preferably computer based) are a music editor's basic necessities. Beyond that, you just need a good pair of ears, knowledge of your equipment, some solid music libraries, and a flexible, creative work ethic. ✺

New Chips on the Block

The latest processors offer more power for desktop productions.

By Brian Smithers

To the desktop musician, life is an endless quest for more horsepower. No matter how fast our computers are, we always want them to chew through audio processes more quickly, provide a higher track count in our DAWs, let us run more instances of our favorite reverb plug-in, and enable our software synthesizers to give us more polyphony. Fortunately, computer processing power is increasing at a remarkable pace.

New generations of both Macintosh and Windows CPUs have hit the market in recent months: Apple released the new G4 series, and PC users can now choose between Intel's Pentium III processor and AMD's impressive new Athlon. Naturally they're faster than their predecessors, but each also offers new features that can greatly enhance music applications.

MUMBO JUMBO

To see what all the fuss is about, I surfed over to the chip makers' Web sites and devoured every press release and white paper I could find. I learned that the new chips sport such practical features as a 9-issue superpipelined, superscalar 80x86 processor microarchitecture and source-synchronous clocking (clock-forwarding) technology. When I read the news about point-to-point topology and high-speed backside L2-cache interfaces, it was easy to see why musicians were getting excited. Okay, I'm fibbing—it wasn't easy at all. To be brutally honest, the only thing I really got out of all that gobbledygook was a naughty chuckle at the word *backside*.

With all due respect to the supremely nerdy, only a few important specifications separate one generation of processor from another. Clock speed is the most familiar. Each new generation of chips supports higher clock speeds than its predecessor; the industry is now starting to measure clock speed in gigahertz rather than megahertz.

But higher clock speed results in only a modest increase in processing power.

STEVE RAMIREZ

It's time for Change...

FIG. 1: Intel's Pentium III processor has increased the capabilities of software synthesizers with its Streaming SIMD Extensions.

To really speed up a computer, you need to change the way that a chip handles data. For example, when Intel replaced its older 16-bit processors with the 32-bit Pentium, the company altered the pipeline so that the chip could perform more operations in parallel (at the same time). Even though the Pentium had clock speeds similar to those of its precursors, it could perform more simultaneous operations and consequently complete the same number of operations in less time.

Cache design is another area in which new chips soar past their forebears. When they were first introduced, Intel's affordable Celeron chips were dogs because they had no onboard cache. Consumers balked at the Celeron's poor performance, so Intel quickly added a small on-die cache that ran faster than the cache on a Pentium II chip. The Celeron's resulting improvement almost rendered the more expensive Pentium II superfluous.

Expanding a processor's instruction set also significantly affects its performance. For instance, Intel's MMX extensions gave multimedia developers new and more-efficient ways to accomplish certain processor-intensive tasks. Applications written to take advantage of the new instructions offered superior performance. Music-application developers were intrigued by MMX's parallel processing, but they were put off by

the fact that both the Pentium and Pentium II processors could not simultaneously use MMX integer math and the floating-point math employed by most audio programs. As a result, MMX is rarely used in music applications. The Pentium III processor avoids this issue altogether by dealing only with floating-point calculations.

NITTY GRITTY

The Pentium III, Athlon, and G4 feature all sorts of improvements that make them faster than their predecessors, and we power-hungry desktop musicians reap the benefits in everything we do. Of all the advancements, however, the new processor instructions are what's bringing the brightest smiles to the faces of developers and musicians. Programmers on both platforms now have access to processor-level functions that speed up those all-important floating-point calculations.

The new instruction set is called Single Instruction Multiple Data (SIMD). For the Pentium III (see **Fig. 1**), Intel has named the feature Streaming SIMD Extensions (SSE). In simple terms, SIMD allows programmers to issue one instruction that will be performed on several different items—it's akin to saying "Feed the pets" instead of "Feed the cat, feed the dog, feed the fish," and so on. SSE applies the MMX's parallel processing to floating-point functions.

Software synthesis is one music application that benefits greatly from SSE. With a Pentium III, Yamaha's S-YXG100 software synthesizer supports twice as many XG wavetable voices as it does with a Pentium II. Its VL virtual acoustic-modeling synthesis is monophonic with earlier chips; with a Pentium III it reaches 8-voice polyphony, due to the newer processor's SSE features and increased overall speed. Other manufacturers also have implemented the Pentium III's SIMD extensions to increase their software synthesizers' polyphony.

As you might infer from its name, SSE facilitates the streaming of multimedia over the Internet; music applications developed for that purpose get a performance boost with a Pentium III. Seer Systems (maker of the *Reality* software synthesizer) offers *SeerMusic Player,* a free browser plug-in that uses SSE to make high-quality multichannel audio available more quickly and easily. The latest version of Beatnik's *Beatnik Player,* a browser plug-in that supports sampling and has a built-in software synthesizer, is also optimized for SSE.

Digital audio mixing and DSP plug-ins eat up floating-point horsepower in a hurry. They, too, stand to benefit from the new processing instructions. For example, Digidesign's recently released digital audio workstation, the Digi 001, relies on the computer's CPU for mixing and effects processing. Whereas Pro Tools has always depended on TDM plug-ins running on custom DSP hardware, the Digi 001 features a new host-based plug-in format called Real Time AudioSuite (RTAS). This means that tons of floating-point operations get dumped on your CPU. Digidesign has

FIG. 2: With its improved FPU specifications and 3DNow enhancements, AMD's new Athlon processor gives desktop musicians a welcome alternative to higher-priced chips.

FIG. 3: Apple's new Velocity Engine–equipped G4 processor delivers impressive performance in software synthesis and DSP-intensive tasks.

SSE-optimized the Windows 98 version of *Pro Tools LE* to ensure that you get the most from your host.

BY ANY OTHER NAME

Advanced Micro Devices has always marketed its K6 processors as a low-cost alternative to Intel chips. However, many desktop musicians have shied away from AMD chips due to the purported inferiority of the K6's floating-point unit (FPU). For the same reason, some manufacturers of plug-ins and software synthesizers still caution their customers to use only true Intel chips.

The latest K6 processor, called the K6-III during development but now known as the Athlon (see **Fig. 2**), decisively resolves this FPU issue. AMD redesigned its FPU for the Athlon, and the results are impressive. The improvements make floating-point calculations cleaner and applications more stable.

The Athlon uses its own version of SIMD enhancements that AMD lumps under its 3DNow banner. (3DNow is a broad category of special chip functions designed to enhance 3-D graphics performance and other multimedia streaming applications.) Because AMD's implementation of SIMD is different from Intel's, software applications have to be adapted specifically to each chip manufacturer's instruction set, increasing the time and cost of the software development process. But now that the Athlon has emerged as a worthy competitor, developers are more willing to cover the extra expenditure needed for adaptation.

Another nontechnical benefit that AMD provides for musicians is the

price pressure it brings to the PC chip market. Although past K6 processors took a backseat to Pentium chips in some respects, their low cost forced Intel to think in terms of value. Consequently, Intel developed the less-expensive Celeron processors, the latest of which have become quite popular among desktop musicians. And because the Athlon can compete with the Pentium III in terms of performance, the bang-for-the-buck factor tips the scales still further in the desktop musician's favor.

BELIEVE THE HYPE

Apple boasts that its new G4 Macs (see **Fig. 3**) are the first-ever personal computers to offer "supercomputer-level performance," based on the new processor's ability to carry out more than one billion floating-point operations per second (gigaflops). The G4 incorporates a new feature called the Velocity Engine. Whereas previous chips dealt with floating-point operations one at a time, the Velocity Engine enables the G4 to handle up to four 32-bit floating-point operations simultaneously.

Maybe it's me, but "Velocity Engine" is one of those Madison Avenue names that immediately make me suspicious. When I first started reading breathless press releases about the new "G4 with Velocity Engine," I thought it had to be the brainchild of the same marketing team that long ago gave us "GI Joe with Kung-Fu Grip." And when I saw that the G4's processing capability is measured in gigaflops, I became extremely wary of the hype.

Fortunately, the Velocity Engine lives up to its publicity. It significantly boosts the performance of floating-point-intensive operations such as synthesis and DSP. For example, BitHeadz has

▼

Digital audio mixing and DSP plug-ins eat up floating-point horsepower in a hurry.

optimized its *Unity DS-1* software sampler and *Retro AS-1* software synthesizer for the Velocity Engine. As a result, both products offer twice their previous polyphony. The improvement also allows more-sophisticated playback algorithms and higher-resolution effects.

As with the Pentium III's Streaming SIMD Extensions and the Athlon's 3DNow enhancements, the G4's Velocity Engine is a boon to digital audio mixing and effects. Version 5.0 of Digidesign's Pro Tools and *Pro Tools LE* have been optimized to take advantage of the new Mac chip's efficiency.

You may not know or care about floating-point calculations, but as desktop musicians we use them constantly. They are a vital part of digital mixing, DSP effects, and software synthesis, all of which are near and dear to our hearts. Luckily, the major chip manufacturers for both platforms are making great strides in speeding up our favorite processes. Any way you slice it, that's music to our ears.

Brian Smithers *is a musician, conductor, and arranger at Walt Disney World. Share your thoughts with him through his Web site at http://members.aol.com/notebooks1.*

Multi-Effects 101

An introduction to the indispensable multi-effects processor.

By John Duesenberry

The early digital effects devices of the 1970s could perform only one processing function. Some years passed before processing power increased (and chip costs decreased) to the point at which a single device capable of generating several effects at once could be offered at a reasonable price. When that finally happened, *multi-effects processors,* which combine delay, reverb, and other types of signal processing, won rapid acceptance. Today, stand-alone reverbs and other dedicated processors are still popular, but multi-effects pro-cessors, which are now available at all price points, dominate the field.

In this article I'll discuss the features that most multi-effects processors have in common. I'll also look at three popular effects that you'll find under the hood of nearly any multi-effects processor: delay, phasing, and reverberation.

EFFECTS PROCESSOR BASICS

The term *multi-effects processor* suggests an independent device, such as a rack-mounted unit. Increasingly, however, multi-effects processors come in other guises. They are built into or added on to instruments or amps, implemented on computer DSP cards, or realized as soft-ware. Nevertheless, most effects proces-sors offer similar effect types and employ comparable signal-routing architectures.

Let's examine how signals enter, travel through, and exit an effects processor. It is common—though not universal—for an effects box to offer a choice of analog and digital I/O. I'll consider the ana-log case first.

Fig. 1 illustrates a simplified mono signal path in which an input signal is fed to an analog-to-digital converter (ADC). The ADC spits out digital samples, which are routed through one or more pro-cessing stages that implement particular effects. The final stage is a digital-to-analog converter (DAC), which converts the samples back into an analog signal.

One of the most critical components of this process is the *sample rate,* which

MULTI-
EFFECTS
101 →

TODAY'S
LECTURE
• PHASING
• DELAY
• REVERB

JACK DESROCHER

determines the unit's frequency response. Nowadays even moderately priced units feature sample rates of 44.1 kHz or higher with oversampling converters, providing CD-quality frequency response. When you consider an effects box, make sure that the specs on the converters are at least up to this standard.

The processor's *sample resolution* (or *word size*) is also important. A higher sample resolution yields a better dynamic range; today's processors usually employ 20-bit or wider ADCs and DACs.

Inside the system, samples are typically represented with an even larger word size. That's because accurate processing of a sample often requires more bits than are in the sample itself. For example, the *internal processing resolution* of an effects box with 20-bit converters is likely to be 24 or 32 bits.

If you're using digital I/O, you don't have to worry about the effects processor's conversion specs because signals enter the system already digitized. But sooner or later you'll have to convert the signal from digital to analog, so you still have to consider the quality of the converters in the signal path before and after the processor. Whether you use digital or analog inputs, the issue of internal resolution is still important.

Most effects processors have left and right inputs and outputs, but some of them sum their "stereo" inputs to mono and then synthesize a stereo field. In contrast, a *discrete stereo* effects processor provides independent signal paths for the left and right channels. Some processors also offer a *dual mono* mode, which applies completely different effects to each channel. In essence, dual mono gives you two processors for the price of one. For example, you could reverberate the left channel signal while applying a phase shifter to the right channel.

Algorithms are the mathematical constructs used by the processor to create effects. For instance, an effects processor might use a single reverb algorithm to create several reverb programs. In a discrete stereo processor, separate mono algorithms are applied to both channels. In a *true stereo* processor, however, a more sophisticated algorithm that accounts for the sound's behavior in a stereo field is applied to both sides.

Algorithms are commonly confused with *patches* or *programs*. A program incorporates an algorithm and a set of parameters (such as delay time, chorus depth, and so on) that modify the algorithm. A program can be either a single effect or an arrangement of processing stages through which a signal passes.

Some programs route a signal in *series* through several stages (see **Fig. 2a**), creating multiple effects; others split a signal into *parallel* paths (see **Fig. 2b**). In many cases, you can tweak the parameters of each effects stage, but you can't arrange the stages in a different order. More advanced multi-effects units let you design your own effects chain. The final stage in many effects processors is a *wet/dry mix* control, which regulates the proportion of processed (wet) to unprocessed (dry) signal at the output.

DELAY EFFECTS

No multi-effects box worthy of the name would be complete without a *digital delay line* (DDL). The basic operation of a DDL is simple. A DDL samples an input signal at a fixed clock rate, reading samples into a RAM
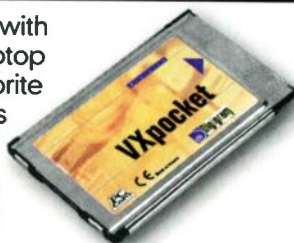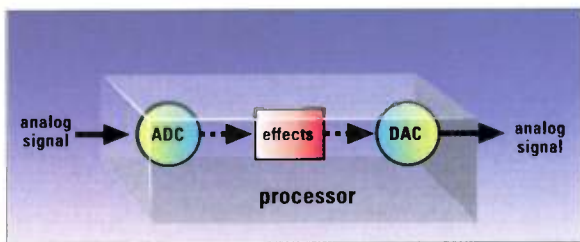
FIG. 1: An analog signal passes through an effects unit in several stages. First is the conversion from analog to digital, followed by the application of the effect, then by a reconversion from digital to analog.

buffer or *queue*. Each sample enters the input end of the queue and moves through it one memory location per clock count. Some time later, each sample emerges at the output.

The *maximum delay* (MD) available in a DDL is a function of the buffer size and the clock rate: MD = buffer size (in samples) × clock period. For example, consider a 32K sample buffer: 1K = 1,024 samples, so a 32K sample buffer holds 32,768 samples. Given a 44.1 kHz clock, MD would equal approximately 0.74 seconds (32,768 samples × 1/44,100 Hz). Thanks to inexpensive RAM, current

DDLs can achieve delays of several seconds or more.

Achieving shorter delays is simple: instead of reading the output from the last location in the buffer, read the output from an earlier one. You select the location from which output samples are read with a software-controlled *read pointer*. If the read pointer for the aforementioned buffer points to location 1,000, the signal is delayed by only 22.7 milliseconds.

The read pointer is often called the *tap*. When you tweak a DDL's delay-time parameter, you're adjusting the tap. A *multitap* DDL has multiple read pointers, allowing you to delay the same signal by different amounts.

Delay in itself isn't an impressive effect; the input signal just comes out later, sounding the same. A notch above this in sophistication is *loop sampling*. In this technique, the DDL records samples into its buffer and

plays them back repeatedly, locking out further input. In effect, the DDL becomes a crude sampler.

The most widely used DDL applications involve feedback, usually accompanied by wet/dry mixing. The simple configuration illustrated in **Fig. 3** yields a variety of audible effects. These can be categorized by range of delay times.

When a signal delayed by about 50 ms or more is fed back into the input, discrete repetitions or echoes occur. The feedback is usually attenuated, which causes the repetitions to fade out. Almost all DDLs offer controls for the *echo repetition rate* (delay time), *regeneration* or feedback level (which determines how long echoing continues), and wet/dry mix.

DDL-generated repetitions are a popular rhythmic device. They're so popular, in fact, that many effects boxes let you set the delay in terms of beats and tempo rather than time, or set the delay time by tapping a footswitch. Many DDLs can also lock the delay time to MIDI Clock messages from a sequencer.

Medium-range delay times of about

10 to 50 ms are often used to produce ambience around a signal. Simply delaying a signal by 10 to 50 ms and mixing it with the original signal can fatten up or "double" a sound. Routing delayed and unprocessed signals to separate channels is a common technique for synthesizing a stereo image from a mono signal (for more on synthesized-stereo samples, see "Master Class: The Splittin' Image" in the May 2000 issue of **EM**). Medium delay times combined with moderate feedback levels produce tightly spaced echoes that add a reverb-like "tail" to a sound.

Complex *comb-filtering* effects occur when a signal delayed by less than about 10 ms is mixed back with the original signal. Each partial of the delayed signal is phase-shifted by a different amount, with respect to the original signal. When mixed, some components are 180 degrees out of phase. These cancel each other out, producing notches in the spectrum. Other frequency components cancel partially or not at all. The resulting spectrum consists of peaks and notches, spaced uniformly through the audio range. When plotted, the spectrum resembles a comb. Comb-filtering effects are reinforced by feedback, which sharpens the peaks.

Flanging is a comb-filtering effect in which the comb is periodically swept up and down the audio spectrum. Flanging requires a *time-variant* DDL; that is, one with which you can vary the delay time by controlling it with some signal. Typical control signals include LFOs, envelope generators or followers, and MIDI continuous controllers.
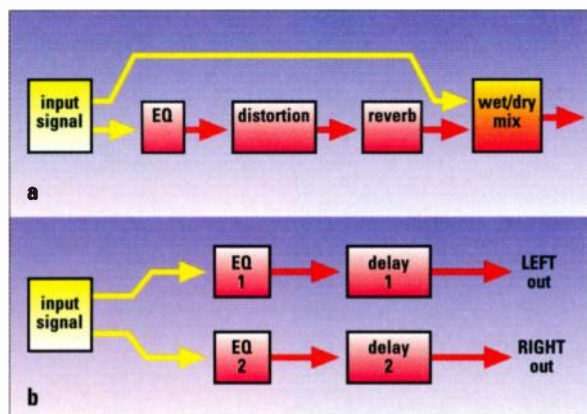
FIG. 2: When a signal is processed in series (a), the output of one process is fed directly into each successive process. When a signal is processed in parallel (b), each of the processes is performed independently of the other.

The classic flanging effect employs an LFO sine or triangle wave to modulate the delay time. The LFO frequency determines the sweep rate, while the LFO amplitude controls the sweep's "depth"; that is, the amount of variation in the delay time.

You can obtain an extension of the flanging effect with a multitap DDL. Routing a signal through a multitap delay while varying the delay time of each tap over a small range is equivalent to running the signal through several flangers in parallel. The result is a complex chorusing effect.

Static comb filtering, in which you don't sweep the comb at all, can also be an interesting effect. A static comb filter acts much like a bank of resonant bandpass filters. Adjusting the feedback amount adjusts the "resonance." Negative (inverted) feedback reverses the spectral peaks and dips.

PHASING EFFECTS

Phasing is another comb-filtering effect, obtained by different means. A phaser routes the input signal through several *allpass filters*. Unlike conventional filter types such as lowpass and bandpass, allpass filters have (ideally) a flat frequency response. Allpass filters have little or no effect on a signal's frequency content. They do, however, shift the the signal's phase. Different frequencies are phase-shifted by different amounts. Thus, when the original and shifted signals are mixed, cancellations produce a spectrum similar to that produced by flanging. Unlike with flanging, however, the spacing, depth, and width of the peaks and notches is adjustable. Sweeping the phase-shift amount with an LFO sounds like a gentler form of flanging. Phasing, like flanging, is often enhanced by feedback.

Analog phasers are making a comeback. For example, Big Briar has just introduced the Moogerfooger MF-103 12-Stage Phaser. And the Electro-Harmonix
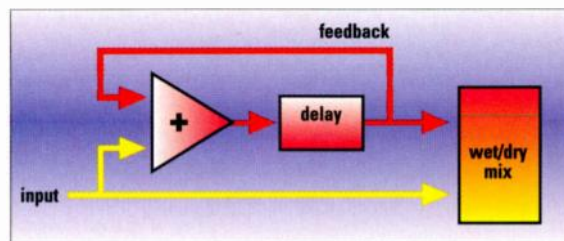
FIG. 3: A delay with feedback mixes time-delayed versions of a signal back in with the undelayed input.

Small Stone phase shifter, beloved of guitarists, was recently reissued.

REVERBERATION

A reverberator usually attempts to reproduce the ambience of an acoustical space; for example, a room, a hall, or perhaps an imaginary space. In a physical space, *direct* sound radiating from an instrument or other source reaches the listener first. *Reflected* sound waves, rebounding off walls and other surfaces, arrive later. *Early reflections,* bouncing directly off the surfaces, are followed by *delayed reflections* that are produced as sounds continue to bounce around in the space. Early reflections are similar to very rapid, but still discrete, echoes. Delayed reflections, often called *global reverberation,* can arrive at rates of 1,000 or more echoes per second, fusing together in the listener's perception. Global reverberation takes some time to build up and fade away.

The space's size and shape, as well as the materials that form its surfaces, determine the strength and duration of the reflections. Reflections in a large hall reach the listener later, and last longer, than do reflections in a small closet. If the space contains materials with a high *absorption coefficient,* such as carpets or curtains, reflections die out more quickly than they would in a room with hard surfaces, such as tile or marble. Moreover, high frequencies are more easily absorbed than low frequencies.

Reverberation is probably the most complex component of a multi-effects processor. The reverb section of a multi-effects box typically combines multitap delay, filtering, mixing, and other processing. These processes can consume a considerable portion of the available computing resources. Therefore, some multi-effects processors (such as the Lexicon MPX 1 and MPX 100) dedicate special-purpose chips to reverb, leaving other hardware to handle more mundane chores.

Because reverb is so complex, most reverberators offer a selection of basic reverb types that imitate the characteristics of an acoustic space, such as "Jazz Club" or "Taj Mahal." Reverb types that mimic electromechanical reverb devices such as plates and springs are also common.

After selecting a reverb type, you work within its parameters to tweak the sound to your liking. The number of parameters you can control and the terms that describe them vary widely. However, programmable reverberators usually provide independent control of three components of reverberation: early reflections, global reverberation, and the ratio of direct to reflected sound.

Early reflections are often simulated by a multitap DDL. The *diffusion* or *echo density* parameter controls the spacing of echoes. *Predelay* sets the amount of time that passes before reflections start. Early reflections may also have a level control and a *shape* or *envelope* that controls the amount of time it takes for reflections to build up and die away. A *high-frequency rolloff* (lowpass filter) is often used to simulate the absorption of higher frequencies. In more sophisticated reverbs, a *crossover* may split early reflections into high-, mid-, and low-frequency ranges, with each range subject to independent level and envelope control.

Global reverberation is controlled with parameters similar to those controlling early reflection: predelay, diffusion, level, envelope, and some type of rolloff or crossover. However, the overall echo density will be much higher than that of early reflections, and the decay time will be longer.

Finally, the relative levels of direct and reverberated signal are set with a wet/dry mix control, by far the simplest component of a reverb.

RELEASE SEGMENT
In this brief introduction, I've only touched upon the basics of multi-effects processing. It's a vast but fascinating subject—the literature on reverberation alone could fill several bookshelves. In a future article, I'll explore many other types of processing that you can find within a multi-effects box. In the meantime, I hope you've been inspired to hit that Edit button and do some exploring yourself.

WRH

A&R Primer

An inside look at the star-making machinery.

By Mary Cosola

Most musicians who know even the faintest bit about the music industry have heard the term *A&R*. Many of them, however, don't understand the full scope of the phrase. It stands for *artist and repertoire*. A record label's A&R department seeks out, signs, and develops acts. The definition is simple, but the work involved is not. A&R plays a pivotal role in the success of artists and labels alike.

The *artist* in the term refers to a musician or band. *Repertoire* is a leftover from the days when most performers didn't write their own music, drawing instead from a body of songs, or repertoire, written by others. "A&R people used to identify artists who seemed capable of making hit records, then team them up with the appropriate writers and songs; hence the *repertoire* part of the moniker," explains A&R specialist Michael Laskow. Laskow is the founder and president of Taxi, a membership-based organization that helps connect unsigned artists with music-industry professionals.

Today the A&R process still works in the same way for many pop, hip-hop, and country acts. But rock artists usually write their own songs, so their labels rarely pair them with songwriters.

The landscape of the music business has changed dramatically in recent years, but record companies still need someone to find new talent, ensure that records are made and promoted, and thus keep the profits coming in. Those tasks still fall to A&R—though with a few new twists.

WHO, WHAT, AND WHERE

Before I discuss how things have changed in the business, let's take a look at the basic functions of A&R. First, what are A&R reps looking for in an artist? Talent, obviously, but talent alone won't land you a record deal. You need to have that "special something."

"A&R people are looking for three

John Bendich is an independent A&R consultant. In addition, he teaches classes in A&R, record production, and songwriting at San Francisco State University.

things when they check out new acts," says Laskow. "First, an artist or a band with a unique vision, someone who sounds different but still has commercial appeal. Next, hit songs. And finally, star quality. If you put any artist possessing those three elements in front of an A&R person, you have a record deal."

John Bendich, an A&R consultant and an instructor in music business and production at San Francisco State University, describes A&R very simply. "The basis of A&R," he says, "is seeing in a performer something that you believe in and going with it."

The primary methods of finding new talent are pretty much what you would expect. Word of mouth is the first and foremost way in which artists are discovered. A&R reps are in constant contact with other industry professionals, such as musicians and club owners, regarding up-and-coming artists. A record label's success depends on its A&R department's ability to find and sign exciting new talent, so A&R people are always on the hunt.

Typically, A&R reps go to clubs to see acts only if they've already heard of the artists. They also attend showcases, although I was surprised to learn that many of the reps don't value showcases any more than they do regular club gigs. "Showcases are a cousin of playing in clubs. It's the same kind of exposure," explains Bendich. "Fewer artists are signed in those situations than through word of mouth. But the positive thing is that if you're playing in a showcase, you already have enough of a buzz in your local market. The labels probably know about you. They rarely see anyone that they haven't heard anything about. At a showcase they get to see how the six bands they've had their eyes on fare in that environment."

A&R reps also find new talent on the Internet. "Some industry people that I know just surf the Internet all day looking for new acts," says Bendich. "It's changing everything. Look at Jimmy and Doug's Farmclub [www.farmclub.com]. The founders of Interscope created that site for new artists. The artists put their songs up on the site, and the public rates them. When the public responds well to an act, the artist is flown to New York for a showcase. The name itself spells it out: it's like the farmclub system in sports."

But having an Internet presence is not enough to get you signed. You still need to put together a high-quality demo, play live as much as you can, and make solid contacts among club owners and industry professionals. The idea is to get your music and your name heard in as many places as possible. Set up a Web site for your band, trade links with other sites, post MP3 files on all the major downloadable-music sites, and follow up with the aforementioned old-fashioned legwork.

What happens once an A&R rep decides to take the risk of signing an artist or a band? It's not as simple as whipping out a contract after a gig and whisking the band away in a stretch limo. First the A&R rep has to convince colleagues at the label that the act is worth the investment. If the rep sees star quality but no one else at the label agrees, forget about it—the deal is over before it started. If the record company is excited about the act, a contract is negotiated. Depending on the genre, the label might then team the artist with songwriters and producers, and the record goes into production. All of these important details take a long time to finalize and put into action.

"An artist who gets signed today won't be known to the American public for one to two years," says Laskow. "Once an A&R rep is interested, it's 30 to 60 days before the contract is even offered, and another 90 to 120 days after that before the contract is done. Then the label starts working on what it's going to do with the artist's or band's career: finding the songs if necessary, hooking the band up with the right producers, making the record, remaking the record, mixing, remixing, doing the artwork, changing the artwork—it's a year and a half to two years before that song has a shot at getting to radio."
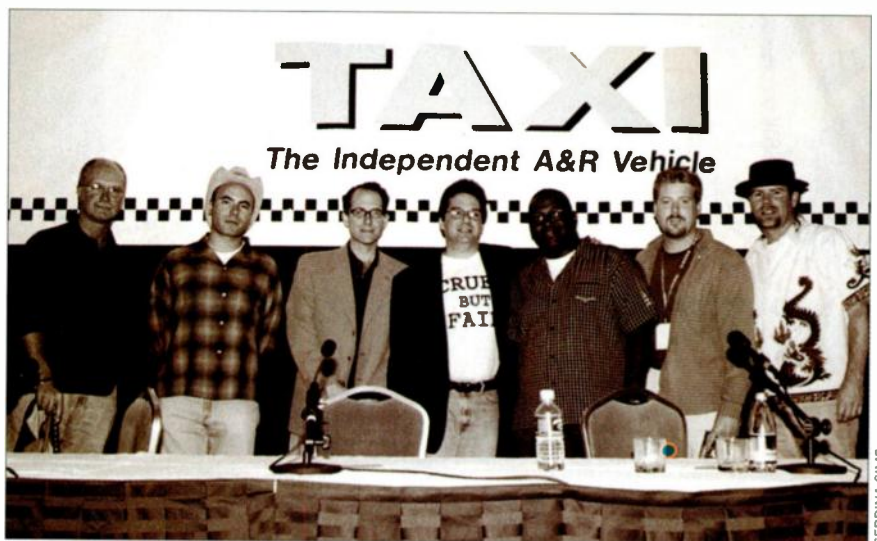
Some major A&R players pose for a photo at Taxi's annual Road Rally convention on November 5, 1999: (from left to right) Tony Ferguson, vice president of A&R for Interscope; Marshall Altman, director of A&R for Hollywood Records; Ron Fair, senior vice president of A&R for RCA Records; Michael Laskow, founder of Taxi; Randy Jackson, senior vice president of A&R for MCA Records; Mike Sistad, director of A&R for Arista Nashville; Andy Olyphant, director of A&R for Almo Sounds.

Simply put, artists who convince A&R reps that they have talent and star quality get record deals; the record labels help them make great records that sell millions of copies; and everyone ends up rich and creatively satisfied. Well, not always, but that's the theory. Now that I've gone over the basics of A&R, I'll move on to the current state of the industry, how things have changed for A&R in recent years, and what it all means to you.

BIG BOTTOM LINE

Record companies were once much smaller operations run by people who loved music and wanted to expose talented musicians to the public. Many small independent labels are now run

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Artist development has all but disappeared.

the same way, but the major record labels seem much more concerned with big profits than with high-caliber music. "The thing that's changed A&R most in the last ten years is the increasingly fast corporate consolidation," explains Bendich. "That has brought a mentality to labels that didn't used to exist—the bottom-line mentality. With the consolidation of all the smaller record companies into corporate conglomerates, labels are now run by people who have no background in music. They're strictly accountants and lawyers; their responsibility is to their shareholders."

The major labels' desire for instant hits is one result of corporate consolidation. In the past, labels developed musical acts. That process included helping the artists develop their sound and refine their appearance, as well as generally nurturing them through the record-making and promotion process. Among the major labels, that practice has all but disappeared.

"The labels don't care about artist development," says Bendich. "It costs a major label half a million dollars to sign, record, produce, and promote an act. So they don't want to hear that it's going to take three to four albums to build that artist. They don't want to wait to see a return on their invest-

ment. They want to make money right then and there. Today the business is based upon a value structure of commodities, whereas before the emphasis was on artistic value. Of course, back then you still had to make enough money to stay in business, but look at Sam Phillips at Sun: he was making money, but he was also making music that he believed in. Those little labels like Sun, King, Chess, and Brunswick were run by guys who loved the music and were willing to take a big risk. They knew that it might take them a while to develop an act, but they believed in that person. That's lost today."

A&R has also been influenced by the public's shifting musical tastes. Modern R&B and hip-hop have dramatically changed the record-making process and, consequently, deal negotiation.

"Hip-hop and the aesthetic that it introduced brought in sampling and an incredibly big shift toward MIDI and the self-contained producer/musician," says Bendich. "That shift crept into all of popular music, and so you started to see the producer who also wrote the songs and was sometimes the artist as well."

The rise of the producer/musician gave labels the economical option of one-stop shopping. Producers face

enormous pressure to churn out hit after hit. Labels sign producers who can guarantee them success, and today's producers often are as famous as the artists they record (and sometimes even more famous). Just look at producers such as Sean "Puffy" Combs, Jimmy Jam and Terry Lewis, and Kenneth "Babyface" Edmonds.

"With the introduction of the superstar producer, artists became interchangeable," says Bendich. "Labels stopped worrying about artist development; they became like factories. Furthermore, most artists in the pop, hip-hop, and R&B genres use more than one producer per album. Labels want to hedge their bets, so they hook the artist up with several producers and hope that at least one track hits the top ten with a bullet.

So where is the love? Where can artists get help with career development? In response to the labels' lack of artist-development resources, many publishing companies have established their own A&R departments. A publisher can bring together artists, writers, and producers on a speculative basis and let them develop the artist's sound, all for much less money than it would cost a label. The team can work on a winning formula, and the publishing company can shop around for a deal. Explains

WHAT NOT TO SAY IF A&R COMES CALLING

I asked Taxi founder Michael Laskow how artists can prepare themselves to deal with an A&R rep. He pointed out that what you *shouldn't* say is as important as what you should say.

"First, when an A&R department expresses interest in your demo, never tell them that they didn't get your best stuff," advises Laskow. "If you tell them you're working on new material that 'will blow our other demo away,' they know that they'll have to wait about six months. They might lose interest by that time. Second, don't tell an A&R rep that your music is all over the map genre-wise. That's a major sin. A&R people want complete, laserlike focus. They can't sell something that's all over the place stylistically. Last, don't go off on long, inarticulate tangents. Be businesslike and to the point. Never waste an A&R

person's time. So many musicians who call Taxi have no idea how to have an intelligent phone conversation. It's sad, really."

Laskow continues with an example of what to say if you're lucky enough to get an A&R person on the phone: "Say, 'Hi, my name is Bob. I have a five-piece band in Minneapolis. We regularly draw 150 people every Friday night when we play different clubs around town. People tell us that our music sounds like Gin Blossoms meets Steely Dan with a female vocalist.' The A&R rep will then have all the information he or she needs to know about that band: where they're from, that they can draw a live audience, and what their music is essentially going to sound like. The A&R person decides whether to continue down the path with them based on those facts."

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Bendich, "Publishing companies have figured out that they can sign an artist who might not be ready today but who has the ability to make it big. Then when the artist gets a deal, that company has the publishing rights."

Laskow points out that many musicians are turning to his organization for the development once provided by record labels. Taxi members submit songs, and the staff screens them. If the screeners think that a song lacks what it takes to pique the interest of an A&R rep, they return the song to

the artist with a constructive critique. If the song does have that "special something," Taxi forwards it to A&R reps, music supervisors, and other parties looking for new material, thus providing the crucial word of mouth that you need to get the attention of the deal makers.

WHAT ABOUT ME?

Much of the responsibility for career development now falls squarely on the shoulders of the artist. Musicians need to have more savvy today than they did

in the past, and they develop that savvy by constantly honing both the creative and business aspects of their acts.

Laskow advises aspiring musicians to broaden their artistic horizons by reading a lot of literature and listening to musical classics. "By exposing themselves to as many different artistic influences as they can," he says, "musicians gain rich, broad sources of inspiration to draw from when creating their own sound."

He continues: "The number-one thing that people can do to prepare for a career in music is to educate themselves about the business of music. It's called the *music business*—that's two words, both very important. Everyone wants to play *Who Wants to Be a Millionaire in the Music Business*, yet no one

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Modern R&B and hip-hop have dramatically changed the deal-negotiation process.

wants to do the homework. The reality is that the people who get discovered are the ones who completely dedicate themselves to building a career. You can't be just a musician anymore. You need to understand the business in order to get signed, stay signed, be successful, and have some money left in your pocket in the end. You need to live it, breathe it, and turn it into an 18-hour-a-day job."

The business of music can be ugly indeed, but it's also very exciting. Why else would so many musicians write, sing, and play their hearts out year after year just for the promise of a shot at the elusive record deal? The opportunity to dedicate their lives and careers to making music is impossible to pass up. The fact that musicians today need to look after their own careers more carefully isn't necessarily a bad thing. If, as Laskow suggests, you turn your passion for music into "an 18-hour-a-day job," you will at the very least gain greater control of your music and your career than most artists did in years past.

Mary Cosola *is a contributing editor for* **EM**.

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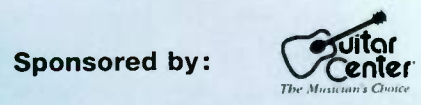
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Each entry must consist of:

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- CD(s) or audio cassette(s) containing ONE song only, five (5) minutes or less in length.
- Lyric sheet typed or printed legibly (please include English translation if applicable). Sheets not required for instrumental compositions.
- Check or money order for $30.00 per song (U.S. currency only) payable to **John Lennon Songwriting Contest. If paying by credit card, $30.00 per song will be charged to your account.**

Entries must be postmarked no later than August 31, 2000.

Please read all rules carefully, and then sign your name in the space provided. If entrant is under 18 years old, the signature of a parent or guardian is required.

1. Each song submitted must be contestant's original work. Songs may not exceed five (5) minutes in length. No song previously recorded and released through national distribution in any country will be eligible. Songs may have multiple co-writers, but please designate one name only on the application. Contestant may submit as many songs in as many categories as he/she wishes, but each entry requires a separate cassette or CD, entry form, lyric sheet, and entrance fee. One check or money order for multiple entries/categories is permitted. (Entrance fee is non-refundable. JLSC is not responsible for late, lost, damaged, misdirected, postage due, stolen, or misappropriated entries.)

2. Prizes: Twelve (12) Grand Prize Winners will receive $2,000 in cash, $5,000 in Yamaha project studio equipment, and a $5,000 advance from EMI Music Publishing. One (1) Grand Prize Winner will receive $20,000 for the "Song of the Year" courtesy of Maxell. Thirty-six (36) Finalists will receive $1,000. Seventy-two (72) Runners-up will receive $100 gift certificates from Guitar Center Stores.

3. Contest is open to amateur and professional songwriters. Employees of JLSC, their families, subsidiaries, and affiliates are not eligible.

4. Winners will be chosen by a select panel of judges comprised of noted songwriters, producers and music industry professionals. Songs will be judged based upon melody, composition and lyrics (when applicable). The quality of performance and production will not be considered. Prizes will be awarded jointly to all authors of any song; division of prizes is responsibility of winners. Void where prohibited. All federal, state, and local laws and regulations apply.

5. Winners will be notified by mail and must sign and return an affidavit of eligibility/recording rights/publicity release within 14 days of notification date. The affidavit will state that winner's song is original work and he/she holds all rights to song. Failure to sign and return such affidavit within 14 days or provision of false/inaccurate information therein will result in immediate disqualification and an alternate winner will be selected. Affidavits of winners under 18 years of age at time of award must be countersigned by parent or legal guardian. Affidavits subject to verification by JLSC and its agents. Entry constitutes permission to use winners names, likenesses, and voices for future advertising and publicity purposes without additional compensation.

6. Winners will be determined by January 15, 2001, after which each entrant will receive a list of winners in the mail. CDs, cassettes and lyrics will not be returned.

I have read and understand the rules of the John Lennon Songwriting Contest and I accept the terms and conditions of participation. (If entrant is under 18 years old, the signature of a parent or guardian is required.)

SIGNATURE _____   DATE _____

WRH

Web Page

By Gino Robair

DOTDOTDOT.COM

WEB SITE OF THE MONTH

So you've finally created your own Web page, including band photos, music files, and links to your favorite sites. However, the fun has just begun: now you have to maintain your site and keep the links current. As a Web surfer yourself, you know how frustrating bad links are. But how can you keep your site neat and tidy without clicking on every link?

The answer is HTML validators, nifty applications that check Web pages automatically. A good online source of validators is the Help4web site (www.help4web.net/links/HTML_Validator.html). This page provides links to dozens of sites offering freeware and shareware validator programs, as well as validation services that can locate broken links, clock a page's load time, test your site for browser compatibility, validate HTML tags, remove syntax errors, check for search-engine readiness, and even check your spelling.

Validators not only help you keep your site up-to-date, but also give you feedback that you can use to build better pages. Interested in learning how your site will be viewed by a Lynx browser (www.delorie.com/web/lynxview.html)? Want to make sure it's accessible to people with disabilities (www.cast.org/bobby)? With the tools listed on the Help4web site, you can perform these and many other tasks to ensure that your Web page continues to be a reliable vehicle for your ideas and music.

When you check out the Help4web validator page, visit its home page, too. There you'll find even more Web-related resources.
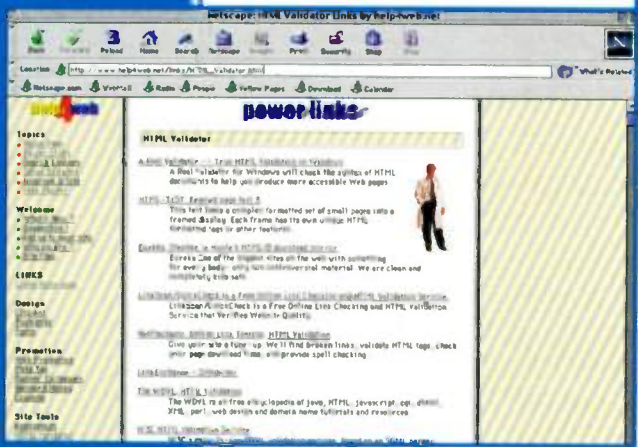
On **Mixman**'s Radio Mixman page (www.mixman.com), users of Mixman *Studio, Studio Mac,* or *Studio Pro* can post their work and listen to and critique the music of others. Visitors view audio selections by artist, title, or upload date, and play them using onscreen tape player–style controls. Minimum system requirements are a RealNetworks G2 player and either Microsoft's *Internet Explorer* 4.0 or Netscape's *Communicator* 4.5. To upload your work, use *Studio Mac* or *Studio Pro* to create and export the necessary Real Audio file . . . **Emagic** (www.emagic.de) and **Rocket Network** (www.rocketnetwork.com) have teamed up with **Harmony Central** (www.harmony-central.com) to promote *Logic Rocket.* This freeware version of Emagic's *Logic Audio* includes RocketPower, the proprietary code that enables a digital audio sequencer to use the Internet recording studios provided by Rocket Network. For this promotion, Rocket Network has set aside public studios that *Logic Rocket* users can access for free. Rocket Network and **Digidesign** (www.digidesign.com) have announced plans to release a RocketPower version of the Pro Tools digital audio workstation. (For more about using the Rocket Network system, see "Web Page: We Have Liftoff!" in the April 2000 issue of **EM**.) . . . At David Emery's **Urban Legends and Folklore** site (http://urbanlegends.tqn.com/library/blhoax.htm), learn the truth about the current rumors and hoaxes spreading like wildfire across the Internet and empty your inbox of all the e-petitions, bogus virus alerts, and other junk-mail clutter.

DOWNLOAD OF THE MONTH

Phil Burke , a developer of interactive music software, has filled an interesting niche in the field of software synthesizers. His newest creation, *JSyn* 13.4 (www.softsynth.com/jsyn), is a cross-platform API (application program interface) for adding real-time, interactive audio-synthesis capabilities to a Web page that incorporates Java. The host-based shareware application gives users the opportunity to explore creative and interactive music-making over the Web in a bandwidth-efficient manner without having to stream or download audio content.

For example, you can use *JSyn* to enhance your Web site with simple sound effects or with complex sound environments that never repeat. The application offers a multitude of useful synthesis components—such as oscillators, filters, and envelope generators—as well as effects, 16-bit sample play-

FIG. 1: The *JSyn* applet *Performance Rack #1* gives you real-time control over a number of parameters.

back, and a programmable scheduler.

To hear what this powerful program can do, download the *JSyn* plug-ins and place them in your browser's plug-in folder. The *JSyn* Examples page includes applets for generating bird and rain sounds, an interactive synthesizer/sequencer called *Performance Rack #1* (see Fig. 1), and compositions featuring *JSyn* by composers Barbara Benary, Nick Didkovsky (aka Dr. Nerve), and Matthew Yee-King.

WEBCAST

Indonesia's indigenous music and the instruments on which it is played are called *gamelan*. The traditional gamelan orchestra includes gongs, drums, plucked and bowed string instruments, flutes, and voices. There are many styles of gamelan music, classifiable by region, ethnicity, and social class. This rich musical tradition has spawned worldwide interest, and gamelan ensembles can be found around the globe. An important source of information about gamelan is the American Gamelan Institute (AGI).

Gongcast (www.gamelan.org/AGI/gongcast.html), hosted by the AGI and curated by composers Jody Diamond, Sapto Raharjo, and I Nyoman Wenten, is a Web site devoted to gamelan music of all styles. Its offerings include tradi-

tional and contemporary music composed and performed by both Indonesians and non-Indonesians, including American composer Lou Harrison.

When you click on the Gongcast button, five sound files (randomly chosen from the site's selections) play. A menu of Gongcast's compositions, titled "What you might hear on Gongcast," appears farther down the page. Gongcast can be heard using *RealPlayer* 3, 5, 7, or G2, the latter two yielding the best results.

multiplexed
downstream
control / data

voice

sprite

audio visual
presentation

2D background

3D objects

multiplexed
upstream
control/data

y

scene
coordinate
system

z

x

user events

audio
compositor

video
compositor
projection
plane

hypothetical viewer

display

speaker

user input

WEB APP

What is MPEG-4?

The Motion Picture Experts Group (MPEG) is part of an international standards organization that publishes specifications for compressing and encoding low bit-rate streams of audio and video information. Each phase of the continually evolving standard is numbered and contains different layers. Thus, MPEG-2 Layer III is phase 2 of the standard, and it contains three layers. It defines the popular music-file format colloquially known as MP3. Perhaps to avoid confusion, the next phase of the standard is called MPEG-4.

MPEG-4 Audio was designed with a broadband, multimedia future in mind. It describes audio "objects" such as speech or music and encodes them using a variety of techniques: Code Excited Linear Predictive (CELP), Harmonic Vector Excitation Coding (HVXC), and Transform Domain Weighted Interleave Vector Quantization, also known as TwinVQ.

The specification creates a standardized file format called Structured Audio that can encode and transmit a variety of synthesis algorithms. The idea here is that it's much

more efficient to send instructions on how to reconstruct the audio on the user's computer than it is to send the audio file itself. A very simple example is, "Play a sine wave at 440 Hz for 1 second." In 30 characters, you have described 176 KB of Red Book audio.

However, producing music requires a more complicated algorithm. MPEG-4 includes a description of the Structured Audio Orchestra Language (SAOL, pronounced like *sail*) and the Structured Audio Sample Bank Format (SASBF), which together describe wavetable synthesis in a way similar to that of DownLoadable Samples (DLS) and Beatnik's Rich Music Format (RMF). Using SAOL and SASBF, a computer downloads and decodes Structured Audio files containing samples, instrument definitions, and performance information. The decoded data is then used to re-create the music on the user's system.

But MPEG-4 doesn't stop there: the standard also defines an interface for Text-to-Speech processes that can be used for lip synching and can modify synthetic vocal quality to take into account factors such as age and gender. Various effects—for example, reverb, echo, and 3-D spatialization—can also be applied to the bitstream, providing a mechanism for synching cinematic-style soundtracks to MPEG-4 Video objects (which are defined elsewhere in the spec).

Because it is designed to cover such a broad range of audio and video applications, the MPEG-4 standard is deep, comprehensive, and extremely complicated. It is also important to remember that it does not define *how* any of the synthesis methods will be implemented. It merely defines a format by which the audio data can be encoded for transmission. The required algorithms are so complex that, unlike MP3 (for which a plethora of free "rippers" is available), MPEG-4 will probably require dedicated hardware to generate the bitstream data (a fact that pleases a number of hardware manufacturers).

In any case, it will be a while before MPEG-4 products reach the general public, although research and development continues in facilities such as MIT's MediaLab. Version 2 of the standard, which adds error-checking and scalability functions, was only released in July 1999. But hold on to your hats, folks—the MPEG committee is already working on the Multimedia Content Description Interface called MPEG-7. Visit the MIT Media Laboratory web site at (http://sound .media.mit.edu/~eds/mpeg4/) for more information on MPEG-4.—*Peter Drescher* ⊛

REVIEWS

EMAGIC

LOGIC AUDIO PLATINUM 4.15 (MAC/WIN)

The rich get richer.

By Mike Levine

The digital audio sequencer market is a potent illustration of the axiom "Competition spurs innovation." The software companies involved are locked in an intense, feature-by-feature battle to produce the dominant product. As users, we're the beneficiaries of this rivalry, and all the major sequencers now give us a dazzling array of MIDI and audio recording and editing options.

No sequencer offers more options than Emagic's *Logic Audio Platinum.* For version 4.0 (4.15 was the current version at press time) the company gave the program a major face-lift, adding a multitude of new features including 31 high-quality audio plug-ins. Emagic tweaked the user interface, too, making this extremely deep and powerful program more accessible, and it also completely rewrote the code for the Windows version to bring it up to speed with *Logic Audio* for the Macintosh. The Mac and Windows versions are now virtually identical, making *Logic Audio* a formidable contender on both sides of the aisle. (I tested the Mac version for this review.)

The most important factor separating *Logic Audio* from its competitors is the degree to which it can be customized to fit individual users' needs. Accomplished

Logic Audio Platinum offers a massive feature set for recording, editing, and mixing MIDI and audio. It also allows a great deal of customization.

users will experience a level of control unmatched by other sequencers. The program, however, does have its own logic (pardon the expression), which takes some getting used to. Due to that fact and to the depth of its features, *Logic Audio* has developed a reputation for being hard to learn. Whereas it's true that becoming a power user of *Logic Audio* can take a great deal of time, getting started is no more difficult than it is with any of its competitors.

A NEW LOOK

If you're accustomed to older versions of *Logic Audio,* the first thing you'll notice about version 4.x is its cooler, more streamlined look (see Fig. 1). The user interface clearly has been designed to be easier on the eyes during long sessions. (For those who prefer the 3.x look, the program can revert: simply check the appropriate box in the Display Preferences dialog box.)

The menu structure is arranged differently, too. The Functions menu now incorporates the former Structure menu, and the new Audio menu gathers the audio features in one place. The reshuffle has made *Logic Audio* more intuitive and much easier to navigate.

A NIFTY ARRANGEMENT

One thing hasn't changed: you'll still spend most of your time at the Arrange window (see Fig. 2). Here you can edit track data, apply quantization, enter point-to-point automation data with the Hyper Draw function, mute and solo your tracks, change instrument assignments, and much more.

Similar to the main screens of other digital audio sequencers, *Logic Audio*'s

Arrange window shows a list of tracks. The track data is depicted in the form of horizontal bars, called Objects, upon which all types of editing can be done. Objects can contain either MIDI or audio data.

By clicking on the telescope icons in the window's upper right corner (or by using the corresponding key command), you can zoom in or out on the tracks

FIG. 2: You'll do most of your work in *Logic Audio*'s Arrange window.

(which can contain multiple Objects), enabling you to view more or less detail on the horizontal and vertical axes. For example, as you zoom in vertically on an audio track, it becomes an editable waveform display. Emagic improved this zooming feature in the new version by making it available for individual tracks in the Arrange window.

Nondestructive MIDI editing is one of the hippest features of *Logic Audio.* It's available through the Parameter box (and the Extended Sequence Parameter box) in the Arrange window's upper left corner; it's also accessible from the individual edit windows. You can quantize MIDI tracks, affect Velocity, change note duration, and alter a number of other parameters in real time without actually changing the track data. A great way to use this feature is to loop a section of a song, then tweak the parameters as it plays back until you achieve your desired results. With *Logic Audio*'s hefty 960 ppqn res-

olution, you have plenty of room for fine adjustments.

Nondestructive editing in the Arrange window isn't limited to MIDI tracks. When editing an audio track, you can add fade-ins and fade-outs and loop individual Objects.

GETTING AROUND

As is its custom, Emagic provides many transport-control choices in *Logic Audio.* In each window, you have the option of showing or hiding a transport-control bar. Or you can open a floating, resizable Transport window that can be customized as to the amount of data displayed. You can also open a second Transport window (resizable as well) and turn it into a large SMPTE position readout.

Because the program lets you open multiple versions of the same window, you can even open a third Transport window and change it into an oversize readout of bars, beats, and ticks. (Actually, the readout shows bars,

FIG. 1: Emagic's *Logic Audio Platinum* 4.x (right) sports a new look that is easier on the eyes than version 3.x (left).

beats, *division,* and ticks. Division refers to the subdivision of the measure. For example, if you set the division option to eighth notes, a readout of 4/1/1/110 would mean fourth measure, first beat, first eighth note, 110 ticks. If you prefer the standard bars/beats/ticks display, you can specify that in the preferences.)

In a Transport window, you can set and enable global looping functions and turn on punch-in/out recording (called Autodrop). You can also turn on the solo function, select destructive or nondestructive recording, and turn the metronome on and off.

In addition, you can use a Transport window to enter tempo and meter information into a sequence. Unlike some of its rivals, *Logic Audio* doesn't have a dedicated conductor track; instead, all meter changes are entered in the Transport window (there is a dedicated Tempo List window elsewhere). Although entering the data is relatively easy, editing it is more cumbersome than it would be in a conductor track. This is one of the few areas in which *Logic Audio* has a slimmer feature set than its competitors. (Emagic recently announced *Logic Audio* 4.2, which will include a time-signature and key-change editor.)

OPEN THE WINDOWS

Although much editing can be accomplished in the Arrange window, *Logic Audio* also offers a number of dedicated editing windows for various functions. As its name indicates, the Event List is an event editor, but it can also be used as a global playlist editor for a sequence, allowing start and end times as well as the length of audio

and MIDI Objects to be edited.

At first glance, the data in the Event List is somewhat difficult to comprehend. The window displays so much information that you must constantly refer to the column headings to understand what you're looking at.

The Matrix Editor, *Logic Audio*'s version of a piano-roll editor, is well designed and easy to work with. Notes—which can be dragged, duplicated, elongated, or shortened—are depicted in different colors, depending on their Velocity; a horizontal line in the middle of each note appears longer on notes with higher Velocities. And you can view multiple tracks in a single Matrix Editor window, a new option.

One of *Logic Audio*'s more unusual windows is the Hyper Editor, which allows you to easily draw and edit Velocity and other controller information for MIDI recordings. The Hyper Editor can also be used as a MIDI drum editor that lets you input individual drum parts on a grid (see **Fig. 3**).

If you prefer to edit musical notes on a staff, check out *Logic Audio*'s Score Editor, a notation editor with enough depth to be a stand-alone program. Not only is it helpful for editing existing parts, but you can also use it for inputting notes in step time. In addition, it offers many features that are specifically dedicated to preparing and printing parts and scores.

Overall, *Logic Audio*'s edit windows are quite functional and well designed. I was a bit surprised to discover that you can't simultaneously view SMPTE times and the bars/beats/division/ticks information in an edit window. You can work around this by opening two of the same edit windows—one for displaying SMPTE and the other for bars, beats, division, and ticks. But that's not as elegant (or as easy to work with) as viewing them both in the same window.
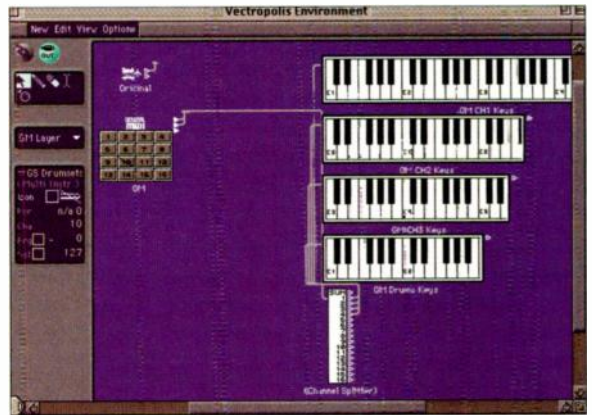
FIG. 4: The Environment window allows you to customize *Logic Audio*'s MIDI implementation.

AUDIO PROWESS

Although nondestructive audio editing, such as cutting and pasting Audio Objects, can be handled from the Arrange window, destructive editing chores are the province of the Sample Editor, a full-featured waveform editor (mono or stereo) that you access by double-clicking on any audio track. It contains tools for drawing out clicks; changing gain, pitch, and tempo; altering the "feel" of your audio files; and many other functions.

One of the Sample Editor's tools is the Time and Pitch Machine, which offers sophisticated algorithms (both conventional and formant-based) for altering audio. An Audio Object's tempo can be changed without affecting the pitch, and vice versa. Both features worked quite well for me, as long as I kept the changes relatively small. (Success also depends a great deal on the source material.) Also available is tape recorder–style pitch-shifting—Emagic calls it *classic pitch-shifting*—in which the tempo changes along with the pitch; it works over a wider range of settings.

The Groove Machine is designed to change the feel of audio tracks. Unfortunately, its controls are confusing, and the manual devotes only a brief section to explaining its use. Through trial and error I was able to convert a drum loop with a straight eighth-note feel into a swing groove, a pretty impressive feat. However, when I tried to alter the feel of a rhythm guitar track, I had little success—this feature seems to work better on peaky, percussive source material.

The Audio to Score Streamer function, reminiscent of the Audio-to-MIDI
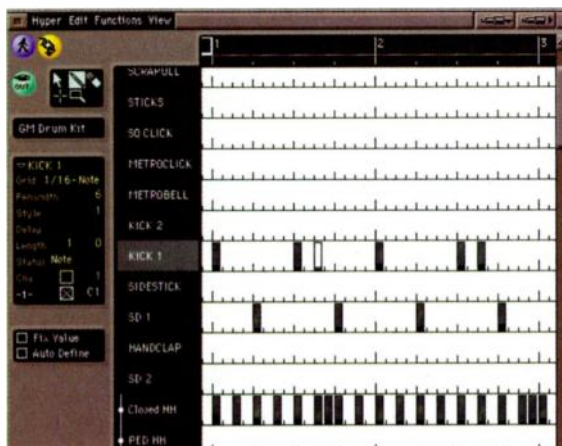
FIG. 3: The Hyper Edit window can be used for editing controller data. It also serves as a MIDI drum editor.

feature in Opcode's *Studio Vision Pro*, does a creditable job of translating monophonic audio files into MIDI files. The simpler the source file, the better the results. On a test vocal track, this function had some trouble tracking the pitches but did well with the rhythms. On a cleanly played guitar track, it worked much better.

SURPRISING SCREEN SETS

With all of the different windows available in *Logic Audio,* navigation could be a nightmare, but thanks to the Screen Sets feature, it's a breeze. You can store up to 99 individual screen layouts in your Autoload (the default sequence that opens when you select New from the File menu); each layout is composed of whichever windows you want. You can switch easily between layouts by pressing numbers on your numeric keypad.

For example, you could have an Arrange window with a Transport window as screen set 1 (the default setting); an Arrange window, Transport window, and linked Matrix Editor as screen set 2; a Hyper Editor and a Score Editor as screen set 3; a Track Mixer on screen set 4; and so forth. At first I couldn't resist the impulse to simply open an edit window rather than go to the appropriate screen set. But once I got used to the concept, it really sped up my work (and decreased the number of open windows on my desktop).

ENVIRONMENTAL CHALLENGE

The window that contributes most to *Logic Audio*'s reputation for complexity is the Environment window (see **Fig. 4**). On the surface, the Environment is where you configure your studio setup. In that sense, it's similar to Mark of the Unicorn's FreeMIDI or Opcode's OMS. (You don't need either program to use *Logic Audio,* but it does support OMS.)

Logic Audio Platinum
Minimum System Requirements
Mac: PPC 603e/120 (G4 recommended); 64 MB RAM (128 MB recommended); Mac OS 7.6.1
PC: Pentium/200 (Pentium II recommended); 64 MB RAM (128 MB recommended); Windows 98; MME-compatible sound card

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$799
upgrade from version 3.5 to 4.x
$199

FEATURES ■ ■ ■ ■ ■
EASE OF USE ■ ■ ■
DOCUMENTATION ■ ■ ■
VALUE ■ ■ ■ ■
1 2 3 4 5

PROS: Offers a powerful and extensive feature set, smooth and stable performance, numerous customization options, and a comprehensive set of audio plug-ins. Has a classy new look.
CONS: Manual is inconsistent. Environment feature is difficult to master. Plug-ins don't offer enough presets.

Performing basic setup tasks is relatively simple, as is setting up custom instruments that trigger a combination of MIDI sound sources.

The Environment is also the place to go if you want to roll up your sleeves and get under the hood of *Logic Audio*'s MIDI engine. Learning how to use this feature—which will take you some time—is similar to learning object-oriented programming. In the Environment, you use "virtual cables" to control the flow of MIDI signals. You can reroute MIDI data and add MIDI effects, such as an arpeggiator, delay, or Touch Track (a feature that allows you to trigger a sequence from a specified MIDI note). You can even set up editors for various MIDI devices.

In *Logic Audio*'s Environment, you can tweak or completely alter almost every aspect of the program's MIDI implementation and signal flow. If you want to use alternate Environments but don't wish to tackle the work yourself, you can simply download or purchase third-party Environments from user-group Web sites. A good number of Environments (including editors for many MIDI devices) are available on the *Logic Audio* CD-ROM.

You'll need the Environment to complete your studio setup's initial configuration, but after that you can ignore it if you prefer. During the review period, I recorded three major projects without once having to tinker with the Environment. But it's always there if you decide to use it, and the potential for customization is almost limitless.

MIX ME UP A GOOD ONE

Not surprisingly, *Logic Audio* offers not just one mixer, but three: the Track Mixer, the Audio Mixer (audio only), and the GM Mixer (MIDI only). Most users will probably opt for the Track Mixer because it lets you mix both types of data from the same screen.

Channels in the Track Mixer correspond to those in the Arrange window. You can avoid cluttering the mixer by deleting any unused tracks in the Arrange window.

The mixer is also where you add plug-ins, either as inserts or by sending to a master effects bus (up to 16 buses can be used simultaneously). *Logic Audio* has its own set of high-quality plug-ins (more on these later). It also supports third-party plug-ins in the VST, VST 2.0, Adobe Premiere, AudioSuite, and TDM formats for the Mac, as well as the VST and DirectX formats for Windows. Automation is straightforward and easy to implement; you can record fader moves, pans, mutes, and even real-time changes of plug-in parameters.

The individual channel EQs are a very convenient part of the Track and Audio Mixers. You can choose from parametric, high shelf/cut, or low shelf/cut, and you can open up to four single-band EQs per channel. Although you could opt for plug-in EQs instead, using too many would put a heavy load on your CPU. The channel EQs are not nearly as processor intensive, yet they still sound quite good. For those attempting to do an entire mix within *Logic Audio,* or for those with only two audio outputs on your sound card, the channel EQs will be a godsend.

During my testing, the Track Mixer suffered periodically from buggy graphics, especially during or after the use of

FIG. 5: The impressive-sounding *Fat EQ* is one of the 31 new audio plug-ins in *Logic Audio Platinum*.

plug-ins. The display texture would change in places, and sometimes plug-in or Transport windows would mysteriously multiply, tile on each other, and change color. Luckily, these erratic graphics never affected the program's musical performance or caused a crash. (According to Emagic, the problem occurs only on Macs running System 8.1 and isn't a concern with System 8.5 and above.)

PLUG-IN CENTRAL

The most dramatic additions to the latest *Logic Audio* are the 31 new audio plug-ins, many of which rival third-party products in quality and features. I don't have the space here to describe each one, but I'll give you some highlights.

Fat EQ (see **Fig. 5**) is the most impressive of the new plug-ins. It features five fully parametric bands that can be switched in and out individually (useful for saving CPU resources). You can select from different EQ curves; your options depend on the band. As you adjust a particular band, the results appear in a nifty-looking bar graph display. More important, *Fat EQ* sounds terrific and compares favorably with expensive third-party EQ plug-ins.

Another standout is *EnVerb*, a reverb that lets you alter the envelope of the reverb tail. You can set up all sorts of gated, reverse, and unusual effects with this plug-in, and it's particularly useful on drum tracks.

Logic Audio Platinum also includes Emagic's flagship reverb plug-in, *PlatinumVerb*. It offers great-quality sound and several editable parameters

including the Spread control, which adjusts the width of the reverb's stereo image. If you want additional reverbs that don't eat up as much CPU power as *PlatinumVerb*, you can open *GoldVerb* or *SilverVerb*; these plug-ins have fewer parameters but will suffice in many situations.

The *Tape Delay* plug-in does an effective job of simulating an analog tape-delay unit. In addition, it lets you set the delay time by clicking on icons representing a 16th note, an eighth note, a quarter note, and a half note; the values are based on the sequence's tempo.

The *Modulation Delay* plug-in offers several effects, including chorusing ("Big Chorus" is a good warm preset), flanging, and doubling. A cool resonator effect (great for those "robot" vocals) is also available. The tremolo preset is workable, but it's not likely to replace your Fender amp anytime soon.

Two more plug-ins are *Overdrive* and *Distortion*. The former does a good job of adding dirt to a signal without sounding too artificial. I tried it on some electric guitar tracks and was able to add realistic-sounding grit as long as I kept the settings subtle. I also had success with the *Distortion* plug-in, which sounded smooth and unmarred by the raspiness often associated with digital distortion.

One of the more unusual plug-ins is *Enveloper*, which gives you control over a signal's attack and decay sections. Its main controls are gain sliders for attack and release. Lowering the release

value reduces a track's audible reverb tail or room ambience. When I tested it on a drum track printed with a healthy amount of reverb, I was able to make the track sound almost completely dry. Raising the attack gain heightens the percussiveness of the audio's attack, but this effect sounds realistic only with subtle settings.

The *Auto Filter* plug-in conjures up filter sweeps and other synthlike effects. I was easily able to transform an acoustic guitar track into a wah-tinged synth sound. Sound designers will have great fun with this one.

Compressor was a bit disappointing. Although it's functional and offers plenty of features, it isn't as sonically pleasing as some of the other plug-ins and can even sound harsh.

For the most part, though, the plug-ins that come with *Logic Audio* are of high quality and cover all the important bases. My biggest complaint is that they don't have many presets, especially compared with what some of their competitors offer.

ADDITIONAL GOODIES

Another major improvement to version 4 is support for 24-bit, 96 kHz recording and playback. The program can also now handle up to 64 audio tracks—either stereo or mono—for each sound card installed in your computer. (*Logic Audio Platinum* supports the use of multiple audio cards, but beware: you'll need a lot of RAM to take advantage of this capability.)

Another noteworthy addition is the ability to record on multiple MIDI tracks. Two options are available: Split mode and Layer mode. In Split mode,

THE LOGIC AUDIO FAMILY

Although *Logic Audio Platinum* is Emagic's flagship sequencer, the company offers three other lower-priced sequencers with varying feature sets. All offer real-time, nondestructive quantization.

Next in line after *Logic Audio Platinum* is *Logic Audio Gold* ($499), which has all the same MIDI features as the *Platinum* version but is a bit slimmed down in the audio department. It has no 24-bit audio support and no TDM support, it supports only

dual sound cards, and it comes with fewer plug-ins.

One step lower on the ladder is *Logic Audio Silver* ($299). It has many of the higher versions' key features, including all of the MIDI editors (Score, Hyper, Matrix Edit, and Event) and several of the audio plug-ins.

On the entry level, *Micro-Logic AV* ($99) is a powerful sequencer for the money, but it has a smaller plug-in set and fewer audio features than its cousins.

incoming MIDI data is sent to its corresponding MIDI channel. This works well for recording several MIDI instruments, each to its own track. Layer mode is good for recording a single part to multiple instruments, because incoming data is sent to all record-enabled MIDI channels.

Also new is the program's ability to open up to nine recent songs directly from the File menu. When you're in a hurry to open a recent composition and you don't feel like searching through your hard drive for it, this is a handy feature.

Logic Audio Environment tweakers will be glad to discover a new feature called Macro Objects. This function allows the grouping of up to 100 Environment Objects.

I should also mention that the user manual has undergone another revision and is now more than 850 pages long. It's generally well written, contains lots of information, and has a decent index. Nonetheless, for all of the manual's detail, information about certain topics (such as the aforementioned Groove Machine) is maddeningly scant.

THE LOGICAL CONCLUSION

Logic Audio Platinum was already a powerhouse digital audio sequencer. With the release of version 4, Emagic made it even better. The plug-ins alone are more than worth the cost of the upgrade; the new look makes a cool program even cooler; and with its cleaner, simpler menu structure, *Logic Audio* is more user-friendly than ever.

Of course, there's always room for improvement. My wish list includes a more consistent manual, an improved compressor plug-in, more presets for the plug-ins, and an integrated conductor track that controls both meter and tempo.

Those are small points, however, and overall this is one impressive product. Between its real-time, nondestructive MIDI editing; its flawless and smooth handling of audio; and its comprehensive mixing features, *Logic Audio Platinum* is an amazingly powerful piece of software. And the more you use it, the better it gets.

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

STEINBERG

LM-4 1.01 (MAC/WIN)

A new VST Instrument to get your groove on.

By Erik Hawkins

Regardless of whether you use *Cubase VST*, you've probably heard about VST Instruments, its virtual instrument plug-ins. *Neon*, a simple analog-modeling synthesizer and the first of these plug-ins, was introduced last year as part of the program's newest releases (version 3.7 for Windows and version 4.1 for Macintosh). VST Instruments are intriguing because they exist entirely inside your computer but function almost identically to external MIDI sound modules. Seamlessly integrated with *Cubase VST*, they can greatly simplify and streamline a computer-based music-production studio.

Imagine never having to deal with another external sound module: no more hassles with plugging in MIDI, audio, and power cables; SysEx dumps; recalling volume settings; tiny LCD screens (or no screens at all)—you know the drill. VST Instruments eliminate these problems by storing everything from signal routings to patches as part of the

Cubase VST session. The flip side of this utopia is that VST Instruments require a lot of processing power. The complexity of the instrument, how smoothly it operates, and how many sounds can play simultaneously are all completely dependent on your computer's horsepower. VST Instruments won't do much for you without a reasonably high-speed system. But computers keep getting cheaper and faster, and the development of VST Instruments pushes on in anticipation of what Steinberg's *Club Cubase* magazine envisions: "1,200 MHz machines."

LM-4 is a straight-ahead drum module (see **Fig. 1**). Its 18 pads can produce up to 18 sounds simultaneously. The drum pads are polyphonic and can be assigned to one of six outputs (one stereo pair and four discrete) in the VST Channel Mixer. Up to eight *LM-4* modules can be loaded per session—assuming, of course, that your computer has enough RAM. (Incidentally, eight is the maximum number of VST Instruments allowed in a *Cubase VST* session.)

Both PC and Mac versions of *LM-4* come on the same CD-ROM, making it convenient for cross-platform users like me. Because *LM-4* works by loading samples from your hard drive into memory, you'll need lots of RAM. Although the plug-in runs on 64 MB, I highly recommend 128 MB or more to take full advantage of sound sets with larger samples. Other than your audio

card, the quality of the samples used is the sole determining factor of how good *LM-4* sounds.

PLUGGING ALONG

Installing *LM-4* is a snap, but you'll need at least 360 MB of hard disk space for the full install, which I recommend. (It includes sound sets from Wizoo that really make *LM-4* shine; although the default kits from Steinberg take up only 17 MB of disk space, they don't do the plug-in justice.)

LM-4 provides 20 drum sets: 10 from Steinberg and 10 from Wizoo. The Wizoo sound sets use 24-bit samples, whereas Steinberg's samples are 16-bit. Wizoo's Acid Jazz XXL kit has some cool brushed toms; the Latin XXL preset contains an excellent, very playable ride cymbal; and the DrumnBass XXL has a fat kick-snare combo. (Wizoo XXL kits include multisampled sounds, whereas economy-size or ECO kits are single-shot sounds; more on both types of kits later.)

To use *LM-4* and any other VST Instruments, you'll need to update to the most recent version of *Cubase VST*. The update is included on *LM-4*'s CD-ROM. Once the software is installed, go to the Panels menu (the Audio menu on the PC version), open the VST Instruments rack, and select *LM-4* as an instrument. Assigning *LM-4* to the Instrument rack is identical to assigning a VST Effect to the Effects rack (see **Fig. 2**).

Clicking the rack's Edit button brings up *LM-4*'s front panel for tweaking. The panel is attractive and nicely laid out, with a look reminiscent of a vintage Linn drum machine. Individual gain and tuning sliders control each pad. Above each pad is a multicolored LED: red indicates that the pad has been triggered, yellow shows that it is selected for editing, and green means it has been

FIG. 1: Steinberg's *LM-4* virtual drum module offers an attractive, clean-looking interface. Each drum pad has a pair of sliders to control volume and tuning.

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FIG. 2: Assigning one or more LM-4 modules to the Instrument rack is the same as assigning VST effects to the Effects rack.

triggered and selected for editing. Once a sound is triggered, the LED glows for as long as the sample sustains. This is a nice touch, providing visual feedback on sample duration. It's handy for all kinds of things, from keeping an eye on polyphony to knowing when to fade.

Like VST effects programs, drum kits are changed or loaded with the program forward and back controls on the rack. As with most plug-ins, *LM-4* parameter settings (tunings, volumes, outputs, and so forth) can be saved for future recall. However, be aware that the Save/Load menu choices in *Cubase* haven't yet been updated to reflect VST Instruments. To save or load a kit's settings, you must use the Save Effect and Load Effect commands. (Settings are also saved within Songs.) The time it takes to load a kit into RAM varies according to the kit's size. It may take several seconds to load a large sound set, but be patient—when the kit is loaded, new names for the pads pop up and it's ready to groove.

With *LM-4* turned on and a kit selected, the Instrument appears in the MIDI tracks' Output menu (where your studio devices show up). Highlight a MIDI track and choose *LM-4* as the output of that track; if your MIDI controller is plugged in, you should be able to play *LM-4*'s sounds. If you don't have a keyboard or other type of controller handy, you can click on *LM-4*'s pads to audition sounds (at a fixed Velocity). However, you can't record that way—after all, this program is a drum module, not a drum machine, and the pads serve other purposes.

ON THE PAD
To tweak a pad's parameters, click once on that pad—its LED should glow yellow. Now you can adjust the pad's volume, tuning, and output. Only one pad can be edited at a time. Volume ranges from 0 to 127. Tuning changes the sample's playback rate, just like on a keyboard. Pads can be tuned up or down by one octave in whole-step increments. I wish the tuning were in cents rather

than in whole steps; that would allow more precise and exotic tunings (such as those used for gamelan or Middle Eastern music).

A big LCD shows the currently selected pad's output. When *LM-4* is activated, all six outputs automatically appear in the VST Channel Mixer regardless of whether you use them. A knob labeled Panorama lets you adjust the pan position on the stereo outputs. It has seven positions on either side of the center. Panning for the individual outputs can be adjusted directly on the VST Channel Mixer.

Two dials in the upper right of *LM-4* are for master or global controls. One dial affects the overall volume for stereo output as well as for the individual outs. The other knob determines Velocity sensitivity for the entire kit: a setting of 0 provides no Velocity sensitivity, while a setting of 10 provides high sensitivity. The Velocity knob is a nice feature; sometimes a kit doesn't need Velocity (for example, with electronic drums), and the master control saves you from having to dissect and reprogram the whole sound set.

Default MIDI note numbers for the Steinberg and Wizoo sound sets range from 36 (the kick) to 82 (usually some sort of percussion instrument). Sounds are mapped out reasonably well; instruments such as toms and hi-hats are grouped together, and kicks and snares are easy to play with your left hand. You can't change MIDI note-number assignments, however, without getting into some text programming, which I'll discuss shortly.

AUTOMATION IN MOTION
LM-4 doesn't offer many parameters, but those that it does provide can all be automated. These include pitch, panning, output assignment, and individual volumes, as well as the master Volume and Velocity dials. Automating these parameters simply requires ensuring that *Cubase VST* is enabled to record SysEx. Then you just pop *Cubase* into

Record mode and move some knobs and sliders directly on *LM-4*'s front panel.

Because fader and pan moves (the most commonly automated parameters) can all be done right on the VST Channel Mixer (see **Fig. 3**), you may not want to automate from *LM-4*'s face. On the other hand, you have to automate *LM-4*'s pitch changes directly because it doesn't respond to Pitch Bend. You can achieve a similar effect in combination with automation by using the pitch sliders, but they change only in whole-step increments. Pitch Bend recognition would be a great addition to *LM-4*. (Subtly changing a sample's pitch during playback with the pitch wheel can be a cool effect.)

One big difference between *LM-4* and an external MIDI sound module is that *LM-4* doesn't respond to patch changes. Because of the time it takes to load a sound set into RAM, patch changes on playback are pretty much out of the question. This isn't an insurmountable problem, though; when you need more *LM-4* sounds, just open another *LM-4* module. (If you run out of VST Instrument slots, it's probably time to start customizing your kits.)

A PAD OF YOUR OWN

It is possible to build your own *LM-4* kit from scratch. You can do it the hard way by programming text files on your own, or simplify things with Wizoo's *LM-4 Drum-Kit Editor*. The free program is available for download at Steinberg's Web site (www.us.steinberg.net). It's in the *LM-4* section of New Products.

Sound-set information is saved as text files that tell *LM-4* which samples and parameters should be loaded. A text file can be opened and edited from any word processor or directly from Wizoo's editor. The Read Me file on *LM-4*'s CD-ROM explains each line of an *LM-4* text file. You can manipulate everything, including pad names, MIDI note numbers, Velocity ranges, and Groups (sounds that share the same voice, such as open and closed hi-hat samples). I slapped together a kit using a bunch of Brazilian percussion samples and was quite pleased with the results. Programming the kit in text took about 30 minutes; using the Wizoo editor cut that time in half.

Like VST effects, sound sets are organized into banks of 20. For example, when you load a Wizoo sound bank, the Wizoo kits in that bank become accessible through the VST Instrument rack's Program buttons. You can rename patches by simply clicking on their names in the VST Instrument rack. I loaded a Steinberg default bank, changed one of the patch names to Brazil, renamed and saved the bank, and then reloaded it. The Brazil kit was then recognized and properly loaded.

LM-4 pads have Velocity zoning, meaning that a single pad can actually hold up to 128 samples, with one sample for each number in the Velocity range. Velocity zoning is an extremely powerful feature that enables you to

FIG. 3: When activated, *LM-4* appears in the VST Channel Mixer, where you can adjust and automate such parameters as volume and panning.

design multisampled instruments like Wizoo's XXL kits. These instruments contain several samples—a variety of hits to paint a more complete sonic picture of the original instrument. One-shot sampled sounds, such as Wizoo's ECO kits and all of the Steinberg drum sets, have only one sample per pad. They aren't nearly as dynamic or as realistic as multisampled sounds, but they're far less memory-intensive. Velocity-zoned sounds can be written into an *LM-4* text file using Wizoo's editor, but be careful: multisampled instruments get large quickly, and without enough RAM you may be unable to load your creation into the VST Instrument rack.

BANG, ZOOM!

Yesteryear's slower computers and early ASIO drivers posed latency problems for virtual instruments like *LM-4*. There was a noticeable time delay (due mainly to the latency of the sound card's ASIO driver) between a key's being struck and a virtual instrument producing the subsequent sound. In the face of such an obstacle, it's nearly impossible to create music effectively. With today's blazing-fast processors and Steinberg's ingenious tinkering, however, latency issues have become a relic: when I played a variety of fast drumrolls and speedy fills in *LM-4*, everything seemed perfectly in time. When I hit a key, I instantly heard a sound.

On my Mac G3/266 MHz computer, playback also seemed particularly smooth. I swear, my dance grooves sounded tighter with *LM-4* than with any of my external MIDI modules. Steinberg claims that *LM-4*'s timing is potentially 40 times better than any MIDI-controlled device's, and I'm a believer.

Initially, I was disappointed by *LM-4*'s limited number of parameters. I like to see filters tied to Velocity sensitivity, ADSR envelopes, pitch envelopes, and other modulation sources. However, I came to realize that these features aren't crucial. In the days of hardware samplers and limited memory, controls like these were used to make single-shot samples sound more natural. With VST effects, comprehensive Velocity zoning, and so much RAM at our disposal, we now have much less need for such things. Still, it would be nice to have them, if only for mad sound designers like me.

CAN'T BEAT THIS

I don't want to sound like I'm pitching an advertisement, but VST Instruments are awesome. Having sound modules integrated into *Cubase VST* is really great—I can't praise this setup enough. Rebooting a session only to find patches, volume settings, and effects returns exactly as you left them is a dream come true. *LM-4* doesn't have all the bells and whistles of its hardware counterparts (filters, ADSR envelopes, tuning in cents, and control by pitch wheel), but these limitations seem minor when you consider *LM-4*'s price, automation potential, and comprehensive Velocity zoning.

Remember, however, that *LM-4*'s sound is determined entirely by the sound set's samples and the quality of your audio card. Don't shy away from this plug-in because you think the Steinberg kits are unimpressive or you don't see enough front-panel controls. Build your own drum set, sample your own sounds, and then—with your custom *LM-4* running through the VST Channel Mixer—choose from a wide variety of VST effects to really mangle your kit. In fact, combined with *LM-4*, VST effects can open a whole world of soundscapes that are perfect for electronica and dance genres. Experiment with flanging snare drums, distorted kicks, and percussion tracks with bpm-matched delays. The possibilities are really quite mind boggling. ⊛

STEINBERG
LM-4 1.01 (Mac/Win) drum module plug-in
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FEATURES ■ ■ ■ ■

EASE OF USE ■ ■ ■ ■

DOCUMENTATION ■ ■ ■ ▮

VALUE ■ ■ ■ ■ ■

1 2 3 4 5

PROS: It's dirt cheap, extremely easy to use, and beautifully integrated with *Cubase VST*. Allows the creation of custom kits with your own samples. Features Velocity zoning for multisampled instruments.
CONS: Lacks filters and ADSR envelopes. Tuning is provided only in whole-step increments rather than in cents.

NEARFIELD MULTIMEDIA

MARIMBA LUMINA

This is not your mother's MIDI controller.

By Larry the O

ooking for a textbook example of a truly slanderous statement? Try this: Don Buchla makes instruments just like anybody else's. Say that in earnest, and no court in the land would refuse Buchla should he decide to bring suit against you. Since the mid-1960s, Buchla has manufactured electronic instruments that are unique in the truest sense of the word. From his modular analog synths, complete with capacitive keyboard, to his first digital synthesizer (the Touche in 1978), to his more recent controller innovations such as Thunder and Lightning, Buchla has long presented the intrepid synthesist with an unsurpassed sonic palette and level of control.

Absorbing and assimilating the depth and character of Buchla's instruments requires substantial study, and a musician must develop new and different techniques to play them. Consequently, large-scale commercial success and the widespread recognition that he richly deserves have eluded Buchla over the years, even as he has grown to legendary stature within the community of the adventurous.

This brings me to the Marimba Lumina, a MIDI mallet controller designed by Don Buchla and Associates and manufactured and marketed by Nearfield

Multimedia, whose business until now has been aerospace technology.

The Marimba Lumina's 5-foot length makes it a bit cumbersome (though no worse than, say, the classic Musser Pro 55 vibraphone), but its 18.5-inch width and 2-inch depth are much more manageable. For all its bulk, the Marimba Lumina weighs a scant 21 pounds. There is no provision for stand mounting, so you'll probably want an X-style keyboard stand.

The face of the instrument contains 3½ octaves of "bars," which are actually 5.63-by-1.75-inch pads silk-screened onto the Marimba Lumina's surface. (All the bars are the same size, as opposed to the bars on acoustic mallet instruments, which decrease in width and length with increasing pitch.) The bars have a basic range from C3 (MIDI note 48) to F6 (MIDI note 89), which can be modified with the transposition functions. The instrument's surface also sports two ribbonlike strips, ten numbered pads, Program and Edit pads, and a 2-line, 80-character LCD. The "black keys" in the top octave are marked with essential control functions (Mute, Solo, Escape, Enter, and Reset) that you use during editing.

The power button and all of the connections are on the right side panel (see **Fig. 1**). Audio outs are on unbalanced TRS jacks. A jack's tip and sleeve carry the output; the ring and sleeve accept signals from an external source connected with an insert cable. This design allows you to sum the output with, for example, another synthesizer, albeit with no independent level controls for the two sound sources. Other connections include MIDI In, Out, and Aux (which currently functions only as Thru); inputs for two switches, an expression pedal, and a trigger; a memory-card slot; and a power supply connector. The power supply employs

a "lump in the line" design. Using an insert cable instead of a regular ¼-inch TS plug in the switch or trigger jacks doubles the number of inputs, giving the Marimba Lumina a total capacity of four switches and two triggers.

The Marimba Lumina uses radio frequency technology to detect the player's actions. The bars emit a weak RF field, and each mallet contains a different value inductor that resonates at its own frequency when brought in proximity to the field. The Marimba Lumina detects this frequency and makes other, more complex analyses to determine not only which bar is struck and how hard, but also which mallet is used and where along the length of the bar it strikes. As a result, the instrument can respond in an entirely different way to each mallet (so that, for instance, each one can send on a different MIDI channel).

INTENTIONS OF MALLETS

You can use only the included custom mallets with the Marimba Lumina. They are color-coded red, yellow, green, and blue. Above the strip on the upper left of the instrument are four LEDs, colored to match the mallets. An LED goes dark when its corresponding mallet strikes a bar, pad, or strip. The four matching lights on the front of the Marimba Lumina (the side that faces away from the player) glow and shine colors over the audience when their corresponding mallets are used. Finally, each bar has an LED that lights up when the bar is struck. Hence the instrument's name—Lumina, meaning *light*.

The mallets themselves resemble Nerf lollipops, being 2-inch foam balls on 15.5-inch plastic handles. They don't have the heft of standard vibe or marimba mallets, but they're not so light as to be unplayable, especially given that you'll never need to strike the Marimba Lumina hard.

For the player of individuated mallets, the repercussions (I won't even apologize for that one) are enormous. Each mallet can play an entirely different sound—or perhaps not a sound at all but a control function. You play the Marimba Lumina as you would an organ or a keyboard synthesizer: to sustain notes, you

The Marimba Lumina MIDI controller, with its myriad options, opens up a whole new world of possibilities to mallet players.

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- **Multiple direct outs** per channel.
- **Optional level to tape** fader control.
- **Assignable, bidirectional MIDI control** of all parameters.
- **Cross patching** allows substitution of channels between various banks.

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FIG. 1: The Marimba Lumina's power button and audio I/O, trigger, and MIDI In, Out, and Aux jacks are on the right side panel.

hold down the mallets on the bars and change volume or tone by sliding them along the bars. The mallets' functions are very broad, giving the Marimba Lumina its remarkable richness.

The unit's audio outputs are for the onboard Yamaha DB51 XG synthesizer, which provides a creditable if unspectacular set of onboard sounds and makes the Marimba Lumina entirely self-contained for many applications. (However, the instrument doesn't have a volume knob, so volume control must be programmed.)

Primarily, though, the Marimba Lumina is a controller, and you can do much more with it by connecting the unit to external devices. It has 50 ROM Programs (which hold Marimba Lumina configurations, not synthesizer sounds) and 50 user memory slots; a memory card can hold 50 additional Programs.

GETTING IN DEEP

Don't be surprised if you feel a little overwhelmed as I explain the Marimba Lumina's software structure. As I said before, Buchla's instruments are unusual and very deep. Because the functions can be so complex, Buchla tends to name and describe them with very academic-sounding terms that, while precise, are not always easy to grasp. Unfortunately, he often uses this type of terminology even when a function is straightforward or the same as one that is commonly found on other MIDI instruments. I suppose that's just another part of the unique conception that makes Buchla's instruments so interesting, but I think the Marimba Lumina would be easier to relate to if common terms were used for common features.

The Marimba Lumina is always in either Edit or Play mode. To call up a Program, you enter Play mode and strike the Prog pad with any mallet except the red one (it has special edit functions). Then you enter the Pro-

gram number, either directly using the numbered pads or with the strip just below the display. You strike the strip's right side to increase the number or strike the left side to decrease it, and you hold the mallet down on the strip to scroll through the Programs. The closer you hold the mallet to either end of the strip, the faster the Programs scroll. (Many parameters are edited in a similar manner—sometimes with this strip, sometimes with the "white keys" in the top octave.) Strike Prog again to call up the Program. You can't hear a Program until you exit this selection process, so it's impossible to audition Programs quickly as you step through them. I most often selected Programs with the strip, because that method shows each Program name as it comes up, whereas using the numbered pads does not.

Striking the Edit pad puts you in Edit mode. Here you make menu selections by striking the white keys of the top octave, which correspond to the menu choices in the display. Values are edited by striking or holding a mallet on the appropriate bar or strip; a single level of Undo is available during editing. Striking the Edit pad again stores the changes and returns you to Play mode. I was disturbed when I received no confirmation message before the Marimba Lumina stored the edited Program: once you hit the Edit pad that second time, your old Program is overwritten, so be careful.

EXPRESSIVE GESTURES

A Program consists of up to 50 Patches, each of which defines a response to a specific action (or *gesture*) performed by the player. A gesture is made up of a *zone* and a *stimulus*—or, put more colloquially, the "where" and "what" of the player's action. The response is defined as an *object type* and its associated parameters. In short, the response is generated when a player does a "what" on a "where."

A zone can be a single bar, strip, or pad; a *cell*, which is a group of those items; or several cells in combination. The Marimba Lumina's surface has eight cells of bars—four black-key cells and four white-key cells—with each cell set to the default size of an octave (the

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split points are user-configurable). Other cells include the numbered pads and the strips. The display uses either text (for individual notes or pads) or blocks (each of which represents a cell consisting of a group of bars) to show the currently selected zone.

A stimulus is an action performed with a mallet, most commonly a strike. Mallet strikes and releases can be separate stimuli, and a stimulus can be defined as coming from any mallet, from a specific mallet, or from any combination of multiple mallets. Similarly, actuation and release of a footswitch can be separate stimuli, as can signals at the trigger input.

Another stimulus (and a good example of the depth of Buchla's instruments) is a *melody strike,* a strike that occurs alone within a defined time window. Obviously, you use this stimulus to differentiate melody notes from chords, but you could also bend it to more twisted applications.

The most common object types are a MIDI note and a continuous controller. In the basic case, the MIDI note number corresponds to the bar that was struck; however, this correspondence can get much more complex with transposition, keymaps, and tuning tables, which I'll get to shortly.

The Marimba Lumina features a wide variety of Velocity maps, for processing the Velocity values, as well as many ways to determine note duration. The latter include fixed; proportional (or inversely proportional) to Velocity; proportional (or inversely proportional) to position on the bar; pitch dependent (the higher the note number, the shorter the duration); terminated by release (the note ends when you lift

the mallet from the bar); damped (the note ends when a second strike at very low Velocity occurs); and terminated by a second strike of any Velocity. You get the idea.

The transposition options are many and versatile. It's easy to assign transposition to numbered pads and switches, thereby expanding the Marimba Lumina's available range.

You can modulate continuous controller values by sliding a mallet up and down on a bar, effectively making each bar a ribbon controller. The values can be sent on two channels—one receiving the original data and one receiving inverted data. (This lets you crossfade two sounds as you slide a mallet along a bar, for example.) Two other interesting options are MIDI controller "LFOs" and, perhaps the coolest of all, the ability to send a controller value proportional to the temporal density (for instance, the faster you play, the higher the controller value that's put out).

Other examples of object types are MIDI messages—Program Change, sequencer Start and Stop, All Notes Off, and so on—and System messages, including operation of the real-time tuning tables. You can send all of this data, which can be defined by a single Patch, on either one or two MIDI channels. I don't have enough space here to go into detail about all of the object types, but suffice it to say that many, many options are available in every case.

You aren't *required* to use all of these fancy features—the Marimba Lumina can certainly do its job in a very simple and straightforward fashion. In fact, the musical use of some of the objects is not always immediately clear. Again,

the richness of Buchla's instruments is apparent in the Marimba Lumina: it gives you two equally useful ways to approach its features. One is to determine the type of control or musical effect that you wish to achieve, and then figure out how to make the unit do it. The other is to ruminate on a feature and devise a musical use for it. It's easy to see how entire pieces can be composed around the Marimba Lumina's capabilities.

IN KEY AND IN TUNE
Two performance features worth singling out are keymaps and tuning tables. Keymaps are among the Marimba Lumina's more straightforward functions: they simply determine the MIDI note that is generated when you strike a bar on the instrument. Keymap 0 isn't editable; it's the default one-to-one mapping (that is, striking pad C5 generates pitch C5, and so on). Keymaps 1 through 3 are user definable.

The tuning tables are more exotic. Each of the ten tables, six of which are stored with each Program, is a sequence of up to 24 notes. Ten presets of common ascending-pitch sequences (chromatic, diatonic, whole tone, pentatonic, and so forth) are available. You can also define a custom sequence of notes in any order. Four real-time tuning tables can be generated by playing

Marimba Lumina Specifications

Audio Inputs	(2) unbalanced ¼" (using insert cable)
Audio Outputs	(2) unbalanced ¼"
Additional Ports	MIDI In, Out, Thru; (2) ¼" TRS footswitch/pedal
Polyphony	32 voices
Sound Engine	sample playback
Keyboard	42 bars
Additional Control Inputs	(2) strips; (10) pads
Internal Programs (ROM/RAM)	50/50
Removable Storage	double-EPROM memory card (space for 50 additional Programs)
Dimensions	60" (L) × 18.5" (W) × 2" (D)
Weight	21 lbs.

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the Marimba Lumina and from incoming MIDI messages.

You can play tuning tables by sliding a mallet along a bar or strip, substituting note messages for what would more commonly be controller messages. Or you can have each strike play a note in the sequence. Real-time transposition can also be applied, so you can modify a scale according to the changing chords of a tune.

You can't play the Marimba Lumina in Edit mode, but its Solo and Audition features allow you to hear what you're programming. With Solo you can play a single MIDI channel; Audition lets you play all active channels, so you can listen to the Patch in context. Just playing while you edit would be simpler, but this is an acceptable compromise.

Buchla collaborator and percussionist Joel Davel has constructed an excellent collection of presets. Some provide useful bread-and-butter applications; for example, with one preset the numbered pads call up different sounds or combinations of sounds. Other presets are more unusual. One, for example, allows you to open and close a filter by sliding a mallet along a bar.

GETTING TO KNOW YOU

I spent a fair amount of time with the Marimba Lumina at Toys in the Attic, my personal studio, and I took it to several rehearsals of Action Palace, my current band. The experience was truly fascinating and enlightening.

My first impression was that I needed to adjust my technique and way of thinking for the Marimba Lumina. When keyboard synthesizers first came out, pianists making the transition to synths found that although the basics of their technique transferred to this new instrument, they still had to deal with significant differences, the most notable being the need for very precise articulation. Guitarists and wind players learning the controller counterparts of their instruments face the same challenge, and so do players of the Marimba Lumina.

For a start, up and down strokes must be cleanly defined. If the sensitivity setting is high enough, the Marimba Lumina will trigger a note when the mallet is very close to the instrument's surface, even if you don't actually strike it. Consequently, when you're playing with four mallets (as

I do) you must hold any mallets not currently in use well above the instrument to avoid false triggers. And because each mallet can be programmed with its own functions, you must hold them in the proper order every time you use them.

Because notes are sustained as long as the mallets remain on the bars, setting the mallets on the Marimba Lumina at power-up or when you're not playing will cause it to play and hold notes. But you can safely set the mallets down on the upper left corner of the instrument.

The Marimba Lumina's sensitivity to a mallet's position on a bar during controller-value generation is another reason that you need deliberate, precise execution. For example, if you hit the instrument too forcefully, you lose the fine mallet control necessary to really tap the Marimba Lumina's capacity for nuance.

You also need at least one set of spare mallets, especially if you regularly perform live. A full set of four mallets is available from Nearfield Multimedia for $200.

Unlike a standard marimba, the Marimba Lumina doesn't have raised bars. Adjusting my technique to accommodate this difference was one of the more difficult challenges I faced. It's a *feel* thing.

The Marimba Lumina can be intimidating at first, but if you make it your ax you'll adjust to all of its differences. You can definitely start using it right out of the box, but expect to spend some time before you feel in command of the instrument and can program it to suit your needs. For me, some programming tasks took a little time to absorb at first but soon became second nature. Learning others, on the other hand, was much harder going.

GIVE ME SOME LIGHT

With a small display (in comparison with a computer monitor), a manufacturer must choose between showing less information with more clarity or showing more information accompanied by cryptic legends. Buchla has opted for the latter—but again, once you are accustomed to them, the menu navigation and odd abbreviations become understandable.

During my review, the Marimba Lumina was a moving target. The manual was going through a major revision at

THE RUMORS ARE ALL TRUE

the time I was testing the instrument and is nearing the end of an even bigger overhaul as I write this. The software is being revised as well. As a result, some rudimentary functions were incomprehensible to me. For instance, I was never able to get the review unit's expression pedal to control the internal synthesizer's volume. You shouldn't run into that problem, though; the Marimba Lumina that you buy will have a very simple parameter for enabling and disabling pedal control of volume. But you may find other simple tasks that are difficult to figure out. (Buchla expects to have finalized the software and manual by the time you read this.)

The manual attempts to be friendly in its approach, but it takes shortcuts, sometimes giving several examples instead of comprehensively listing options. In a number of instances, my efforts to decipher features were hampered by unclear documentation. Overall, the manual adequately conveys the Marimba Lumina's concepts and features, but it's not as successful at describing the details.

STRIKINGLY UNIQUE

The Marimba Lumina is an instrument just like anybody else's—if anybody else is like Don Buchla. It is extremely powerful and capable of levels of expression unmatched by anything that Buchla didn't build. It also is quirky, takes substantial effort to understand and master, and requires you to adjust your mallet technique. The internal synthesizer is quite serviceable though not outstanding; I suspect that a player can do much more using an outboard source such as a sampler or a versatile sound module.

The unit's LEDs provide a rad little light show. They combine with the Marimba Lumina's appearance and the Nerf-like mallets to create a very theatrical look.

Don't consider the Marimba Lumina if you want an instrument like everybody else's. This is a new instrument that holds incredible possibilities for mallet players, as well as for those who don't know a marimba from a glockenspiel. Is it cheap? No. Is it easy? No. Is it worth it? Hell, yes!

Larry the O plays MIDI mallets with Action Palace and provides a variety of music- and audio-related services through his San Francisco–based company, Toys in the Attic.

WRH

S S E Y O

KOAN PRO 2.2 (MAC/WIN)

Computer-generated music for work and play.

By Allan Metts

Computer-generated music has been with us for decades. Research labs have been developing algorithmic software for years, and the commercial products are starting to show a few gray hairs. I'm happy to report that after all this time, the state of the art for computer-created music is still advancing nicely—and if you want proof, take a look at SSEYO's *Koan Pro* 2.2.

Make no mistake: *Koan Pro* is no ordinary auto-accompaniment program. It won't have you typing in chord progressions, building verse-chorus-bridge structures, or jamming with your computer into the wee hours of the morning. *Koan Pro* is a generative music-making program. You give it various parameters to work with, and it in turn gives you music. *Koan Pro* generates music in many different ways, so you can use it to create anything from ambient textures to dance beats.

KOAN HOME

First, let's get a little terminology out of the way. Each composition that you create with *Koan Pro* is called a *Piece*. Pieces are composed of one or more *Voices*, which come in several types. A Voice is a single "player" in the *Koan* system. It plays one Instrument (that is, one patch), which can be an MP3 or WAV audio file, a SoundFont, or a MIDI file. A Voice is somewhat similar to a track in a traditional MIDI sequencer. Instruments, envelopes, parameters, and rules determine exactly what a Voice will do (more on this shortly).

As it starts up, *Koan Pro* offers you a choice of *Templates*. A Template is a "head start" for your *Koan* Piece; Template options include Normal (which brings up a simple one-Voice Piece), Dance, Latin, Rock, and others. Most of the Templates give you nothing more than a good drumbeat to start with.

Koan Pro's user interface (see **Fig. 1**) holds dozens of parameters. Each Piece has its own window, and you can load multiple Pieces simultaneously. A window consists of rows, one for each Voice in the Piece; each row contains the Voice's parameters and envelopes.

(As their name implies, envelopes alter parameters over time.)

Row and column sizes are adjustable, but some autopositioning options for the windows would have been useful. As I worked with the program, I found myself constantly adjusting the sizes and positions of the Piece windows and the controls inside them.

You can reconfigure elements on the screen by clicking on the numerous View buttons in each window. Fifteen buttons enable different parameters, and six others enable different envelopes.

So what are these envelopes? They control a Voice's volume, pan, Velocity, and Velocity change over the duration of the Piece. (Velocity change defines how much a note's Velocity can differ from that of the previous note.) Various drawing tools allow you to create any envelope shape imaginable, so it's easy to make a constantly changing Piece in which Voices fade in, fade out, and move around the stereo field.

TYPE-O-MATIC

Koan Pro provides six Voice types, each with its own special powers. The Rhythmic Voice is good for generating musical phrases. With this Voice type, you set minimum values as well as range values (that is, how far above the minimum a parameter value can be) for pitch, for phrase length, and for phrase-gap length, which is the amount of time elapsed between phrases. You can also have rests occur within phrases a certain percentage of the time.

The Ambient Voice is good for drones, pads, and special effects. It isn't necessarily tied to bar or beat boundaries. Instead, it plays notes for a certain duration (expressed in seconds, milliseconds, or 60ths of a beat). You can specify minimum and range values for duration and gap length.

The Follow Voice is intended to harmonize or echo other Voices. You identify the Voice that you want it to follow (called the *lead Voice*) and, using a percentage value, indicate how often the
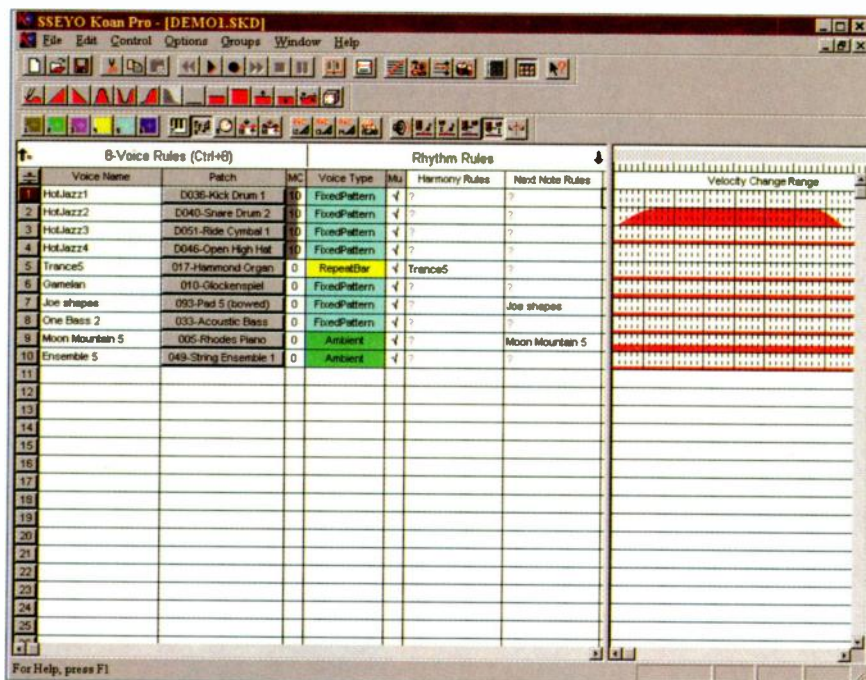
FIG. 1: *Koan Pro's* main screen provides access to all of a composition's parameters. Many of the parameters on the left side of the screen can be controlled by the envelopes on the right. Different views give different sets of parameters.

FIG. 2: The Melodic Pattern Editor, shown here, is used to edit Fixed Pattern Voices. It's handy when you want *Koan Pro* to play specific melodic phrases with little or no randomization.

The Repeat Bar Voice allows you to establish repeating themes in your music. You decide which Voice should be repeated and set the likelihood of repetition as a percentage value. Whenever *Koan Pro* determines that a repeat should occur, it uses the minimum and range values of the Repeat Bar History and Repeat for Bars parameters. Repeat Bar History indicates how far back in the Piece *Koan Pro* can search for the Voice that it needs to repeat; Repeat for Bars determines the length of the repeated passage.

The Listening Voice responds to input from an external MIDI source, such as your sequencer or controller. Thus, you can play along with the *Koan* Piece, as well as harmonize, delay, repeat, and note-correct the external source and the other *Koan* Voices. A Listening Voice has parameters for setting the range of "listened to" notes and for adjusting the notes or note Velocities as they come in.

FIX IT

The final Voice type is the Fixed Pattern Voice. You'll use it when you want

effect should occur. Follow Voice delay values (both minimum and range) are expressed in seconds, milliseconds, or 60ths of a beat. You can create some interesting imitative effects with delay times of four or eight beats. A Follow Voice's shift interval, which also takes minimum and range settings, determines the interval that will be used to harmonize the lead voice.

the Voice to play from a set of predefined musical phrases. In *Koan Pro* such a set is called a Pattern, and a phrase within the set is called a Sub-Pattern. The Fixed Pattern Voice is good for drums and repeating bass lines; for example, you might use one to create snare drum hits every second and fourth beat.

You edit Patterns in *Koan Pro*'s three Pattern Editors. The Pattern Editors give you access to the Patterns' Sub-Patterns, and you can set the probability that a particular Sub-Pattern will be played.

To create a Sub-Pattern in the Melodic Pattern Editor (see **Fig. 2**), you specify a root note, scale, and number of bars. Then you draw the notes into a display that has notes and beats organized in rows and columns (the notes are restricted to those in your chosen scale). Many standard mouse operations, such as dragging, event copying, and lassoing, can be used to tweak your data. You can edit the Velocities of the individual events and import or export MIDI sequences.

The Rhythmic Pattern Editor, which contains only one row, is used when you want to specify just the rhythm to be played (*Koan Pro* chooses the notes according to parameters that you set elsewhere.)

With the Pattern Sequence Editor, you can limit the selections that *Koan Pro* uses. For example, you could ensure that Sub-Pattern 1 gets played first, followed by two iterations of Sub-Pattern 2, followed by one iteration of Sub-Pattern 3. You can even set up several different Sub-Pattern sequences, assign probabilities to them, and then let *Koan Pro* determine which one to use.

A couple of important points need to be made. First, *Koan Pro* transposes your Pattern to other root notes as part of its normal transformation functions (although the Force Frequency button can prevent this). Second, Patterns are monophonic—that is, they can play only one note at a time. If you want to use chords, you have to set up Follow Voices or use Chord rules. This limitation is unfortunate, though no doubt necessary to ensure that Follow Voices can correctly track Fixed Pattern Voices. The downside is that if drum notes need to sound simultaneously (bass drum and hi-hat, for instance), you can't create the drum part in one Pat-

tern Editor—you have to spread it out across multiple Voices. If chords were allowed, the program could function as a full-fledged pattern-based sequencer. (It's still a pretty good one anyway, as long as you don't mind monophonic tracks.)

Fixed Pattern and Repeat Bar Voices can mutate over time. You specify the likelihood that one or more notes will mutate, and you establish a range of bars over which the mutation can occur. For instance, you can set a 15.5 percent chance that a note will change

every five to eight bars. Once a Pattern (or Repeat Bar) mutates, it will not revert to its original form. This allows the Piece to evolve over time.

Finally, Fixed Pattern Voices can have their own meter, creating interesting possibilities for polyrhythms and other such goodies. You can also set the likelihood that *Koan Pro* will use the Patterns at all; for instance, you might program it to play the Patterns 60 percent of the time and make Pattern-playing random for the remaining 40 percent.

OBEY THE RULES

Koan Pro offers quite a bit of note-generating power. However, such a program would be nothing but a noise-maker if it only spit out random notes in any order and at any rate of speed. To make music, you must have rules.

Koan Pro has four types of rules. Scale rules define which notes the program will choose from when it looks for notes to play. Among the default Scale rules are major, minor, and pentatonic, and you can also create your own with relative ease.

An option in the Scale Rules window (see **Fig. 3**) lets you establish the probability that a given note will sound. For one Voice, you may want to weight all of the notes in a blues scale equally. But for another Voice, you may want to give high probabilities to the first, fourth, and fifth scale steps and assign medium-high probabilities to the third and seventh.

Harmony and Next Note rules work much like Scale rules. Harmony rules control the likelihood that a note will be chosen, relative to the harmony implied by the currently sounding notes. Next Note rules establish the probability that the next note after the Voice's currently sounding note will be chosen. Using Next Note rules, you can set up melodies that move from note to note only in half or whole steps, or you can create melodies that jump around radically.

Rhythm rules aren't really what their name suggests (I'd call them "Note Length" rules). With these, you establish the probability that notes of a par-ticular duration will sound. If you choose high-probability settings for whole and half notes, your Voices will play long, sustained notes. High probabilities for eighth and 16th notes will result in fast passages. One of the Rhythm rules makes your Voices play in a syncopated fashion.

CHORDS AND CONTROLLERS

As a default, each *Koan Pro* Voice type will generate monophonic phrases. If you'd like to generate chords instead, head to the Auto Chord parameters. Here you can input minimum and range settings for chord depth (how many notes *Koan Pro* will play), interval (how far it will go for the next note in the chord), and chord delay. Longer chord delays produce arpeggios instead of chords.

You also have your choice of chord "strategies." You can tell *Koan Pro* to consider the Scale, Harmony, and Next Note rules, just the Scale rule, or no rules at all. You can set the probability that a given note in a chord will sound, and you can vary the Velocities of the notes in a chord; these two features help make the chords sound much more natural.

Koan Pro also boasts hefty real-time controller support. For each Voice, you can set parameters for several MIDI continuous controllers, as well as a number of the "special" controls (for example, envelope, LFO, and filter settings) on Yamaha XG, Roland SCC1, and Creative Labs AWE devices.

You can do even more with Pitch Bend, Modulation, and Volume data.

FIG. 4: The Windows version of *Koan Pro* includes the *Koan File Player* editor and player, which assembles collections of Pieces. The player supports all types of *Koan* files.

You can specify minimum and range values for these messages, and set how often (again, with minimum and range values) you want any changes to take place. The Modulation values even have separate "pulse" parameters that dictate how long *Koan Pro* applies the controller.

You can have two user-definable MIDI controllers, which have similar capabilities. They can also behave like LFOs. In addition, you get a built-in ADSR envelope generator, a fixed pitch offset for tuning (this uses the Pitch Bend control), and a "micro" note-delay feature for rushing or dragging the beat. Whew! No lack of control here.

SOUND OFF

What sounds come out when *Koan Pro* generates notes? You have many choices. Each Voice has a patch parameter that can be set either to a General MIDI patch number or to a patch-and-bank number combination for Roland GS or Yamaha XG support. Of course, you can use an external MIDI synth as well, but you can't import patch names into *Koan Pro*.

The program also supports Sound-Fonts. You can browse through a SoundFont file, choose an Instrument, and have *Koan Pro* load it into any SoundFont-compatible card before playback starts.

As if that weren't enough, you can also use MP3 or WAV audio files as a Voice's "noisemaker," provided the appropriate Windows or Macintosh drivers are installed. The audio file's frequency can be fixed (for ambient noises or special effects), or the file can be pitch-shifted across the keyboard.

FIG. 3: Scale rules determine which notes each Voice is allowed to play. In this screen, you can create new Scale rules and establish the probability that each note will play.

KOAN PRO

You can even set up keyboard splits using multiple audio files, but you'll need to use a complex text syntax for this.

"Audio breakbeats" can be created by using a collection of numbered audio files (such as those generated by a product like Steinberg's *ReCycle* loop editor). *Koan Pro* will assign these files to consecutive MIDI note numbers, which can then be used in a Fixed Pattern or other type of Voice.

MAKE A LIST

Every *Koan Pro* parameter can take a list of entries instead of a single value. Each time you play a Piece, the program will choose new values from the lists. Given the multitude of parameters, your Piece might be drastically different each time you play it. You can also use the list feature to change a Voice's patch every few bars.

Koan Pro's parameters can be randomized. You can create entire random pieces from scratch, make subtle changes to an existing template, or simply randomize a few select parameters. The program gives you plenty of control over how much randomization takes place; for example, you can choose the percentage of parameters that change, what types of parameters change, and how much they change.

If your Piece is already close to being what you want, you can mutate a

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SSEYO
Koan Pro 2.2 (Mac/Win)
music-generating software
$199.95

FEATURES	■ ■ ■ ■
EASE OF USE	■ ■ ■
DOCUMENTATION	■ ■ ■ ■
VALUE	■ ■ ■

1 2 3 4 5

PROS: Deep program offers many ways to generate music. Supports SoundFont, MP3, and WAV files. Provides a good vehicle for music on Web pages.
CONS: Program is difficult to learn. Documentation is wordy and unorganized. Support for song structure and chord progressions is limited.

parameter rather than randomize it. The operations are similar, but mutation makes random changes to the current value, whereas randomization chooses new values from scratch. Both of these functions are great for generating new ideas and getting the creative juices flowing.

THE SOUND OF KOAN

What does *Koan* music sound like? You can use it to create beats for everything from straight rock to modern jazz. To my ears, however, *Koan Pro* is best at generating complex, evolving textures. The program could crank out months of music for a shopping mall or a new-age bookstore.

Koan Pro's attempts at more upbeat music are not quite as impressive, although I was able to create some decent techno dance Pieces. Other Pieces containing nothing but percussion sounded fine—in fact, the program is an excellent session drummer that can come up with its own new ideas. However, the music on top of the drumbeats tends to sound a little chaotic. Popular music requires a certain structure in its chord progressions and repeated passages. *Koan Pro* generates music that can evolve over time, but it doesn't currently "think" in terms of verses and choruses. (The manufacturer claims that the forthcoming version of *Koan Pro* comes with enhanced features for creating dance material.)

Koan Pieces are particularly well suited for use on Web pages. Instructions and JavaScript examples on the SSEYO Web site show you how to post *Koan* Pieces on your own site. The company provides a plug-in, *Koan Plugin* 6.1, that allows visitors on your site to hear your *Koan* music. When they click on a *Koan* file on the site, they can download the plug-in and listen to your material. And because *Koan* Pieces can constantly evolve, the music need never become repetitive or tiresome.

The program's built-in recorder can record *Koan* Pieces as MIDI or WAV files. The WAV recorder supports 8-bit and 16-bit recording at several different sample rates and can record your sound card's synthesizer output straight to disk (provided your hardware supports this feature). Also included in the Windows version only is the *Koan File Player* (see **Fig. 4**), which is installed separately. This package lets you create collections of *Koan* Pieces and play them one after the other.

Koan Pro's documentation is extensive, but it is provided entirely in HTML format; there is no printed manual. This is a very complex program, and good documentation is essential. The online docs taught me what I needed to know, but I often had to read through more than I *wanted* to know. A little less wordiness and a little more organization would help tremendously. SSEYO says it will include new tutorials in a future version. The company also tells me that the next *Koan Pro* version will have even more goodies, such as a built-in software synth and enhanced MIDI-file capabilities.

All in all, *Koan Pro* is a huge program with a stupendous amount of music-making potential. If you're looking for new ways to generate music, if you want to add sound to your Web site, or if you need some new creative ideas, you'll like what you can do in *Koan Pro*.

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TECHNOSAURUS

MICROCON AND CYCLODON

Portable analog synth modules that are real monsters.

By Gino Robair

In a radical departure from its high-end Selector-series modular synthesizers, Technosaurus has recently introduced its Small Monster line of discrete analog synth components. Although the Small Monsters are aimed squarely at the groove and dance markets, these new instruments offer a host of cleverly implemented features that many synth enthusiasts will find useful.

The first two models in the series are the Microcon, a monophonic analog synthesizer (see **Fig. 1**), and the Cyclodon, a 16-step analog sequencer (see **Fig. 2**). The features of the Microcon and Cyclodon are complementary, making the two devices a well-matched pair.

POCKET MONSTERS

Technosaurus designed this series with portability and ease of use in mind. Each of the instruments is roughly the size of a VHS videocassette. Such a small footprint means that they can sit on the little shelf above a Minimoog keyboard, or perch on the unused portion of a keyboard controller. For the musician on the go, a couple of these modules and a small drum machine will fit neatly into a compact suitcase.

Each of the instruments is lightweight and housed in a sturdy metal box. Rubber feet on the bottom keep the units from slipping around as you tweak the knobs. In fact, the knobs are also rubberized and provide a nice amount of resistance as you work with them. This helps reinforce the feeling that the Microcon and Cyclodon are built to last.

Each of these instruments has I/O jacks on the right side. Unfortunately, they both require a wall-wart power supply—in this case, 12 VDC. (Technosaurus claims that you can power two units using a single power supply and a Y-cable.) The option of using batteries to power the modules would have been nice; the wall warts that I received with the units are almost half the size of the instruments themselves. In addition, the power jacks on both the Microcon and the Cyclodon are located at the bottom of the right panel instead of at the top; because I wanted the controls to be closer to me than to the wall warts, I had to weave the power lines under all of the other cables. This proved to be a tricky undertaking in some of my instrument configurations.

ONE VOICE, NO WAIT

The Microcon is about as simple to operate as any monophonic synthesizer I've seen. Once you power it up, you can plug it into an amplifier, press the Trigger button, and be on your way. As with the earliest analog synths, you can't use the Microcon to save and recall a patch. Whatever sound you have dialed up on the front panel is what you'll hear the next time you turn it on.

The Microcon's front panel is divided into six sections—Control, VCO, LFO, Envelope, VCF, and VCA—and includes 12 knobs, 7 switches, and 1 button. At the panel's far left is the Control section, which includes the master tuning control, a button for triggering the envelope generator, and a Glide knob for setting the amount of portamento between notes.

Next is the VCO, which offers two waveforms: square or sawtooth. You can also mix in a square-wave oscillator pitched an octave below using the Sub-Mix-VCO knob. The VCO can be transposed down two octaves using the Octave switch. The final control in the section is an LFO Mod knob that controls the amount of LFO modulation added to the oscillator.

There are only two controls for the LFO: frequency and wave shape. As with the VCO, you have a choice of square or sawtooth waves. The frequency of the LFO is controlled by the Speed knob and ranges from subharmonic (roughly one cycle for every 15 seconds) well into the audio range. LFO modulation can be added to the VCO as well as the VCF.

Acting primarily on the VCA, the envelope generator gives you continuous control over attack and decay times. A third envelope parameter, Release, can

FIG. 2: The Cyclodon allows you to change the number of steps in the sequence in real time.

FIG. 1: The controls on the front panel of the Technosaurus Microcon are well placed, with 12 knobs and 7 switches fitting into its tiny footprint.

be switched on or off. The release time is the same as the decay time, but by having Release switched on, you get the entire envelope if you press and let go of the trigger button when the VCA is in AD mode.

The Microcon also includes a good-sounding lowpass VCF. You're given continuous control over the filter's cutoff frequency, the resonance, the amount of LFO modulation, and the amount of control that the envelope has over the filter. When the CV switch is in the On position, the filter is controlled by the signal at the CV input. The Mode switch determines whether the filter's frequency cutoff is 12 dB per octave (for a 2-pole filter) or 24 dB per octave (for a 4-pole filter). Of the two settings, the 24 dB cutoff sounds the smoothest.

The main feature of the VCA is the AD/Gate switch. In AD mode, the attack and decay settings of the envelope determine the volume contour. In Gate mode, the volume is switched on by the gate, and the envelope settings have no effect on the volume of the signal.

In addition to the wall-wart power jack mentioned earlier, the Microcon has a single ¼-inch output jack and ⅛-inch minijacks that accept CV and Gate signals. The Gate input requires a stereo miniplug; the CV input can accept a mono or stereo miniplug, depending on the application.

A rundown of the Microcon's features sounds more complicated than it really is. The good news is that someone without any previous knowledge of analog synthesis can get good sounds right away with this unit.

TUNING IN

As you would expect with a modern discrete analog circuit, the Microcon's

VCO pitch is solid. With the VCO's mix knob turned fully clockwise and the tuning knob at 12 o'clock, the pitch is A-440 Hz (just as the manual specified).

The tuning knob gives the Microcon's VCO a range of a few cents less than an exact tritone (±6 semitones) up or down from the center detent. The manual, on the contrary, claims a pitch range of a fourth (5 semitones) above and below the center detent. In addition, I found that the octaves of my particular unit were not completely in tune. This was most apparent at 4- and 5-octave leaps when I was playing the Microcon and Microcon II (see **Fig. 3**) in unison from a keyboard as a lead instrument. I also noticed that switching the Microcon's oscillator down two octaves (the –2 position of the Octave switch) lowered the pitch a few cents beyond two octaves.

Depending on your point of view, the imperfect octaves could be a blessing or a curse. As part of the dual-voice lead instrument mentioned above, the subtle octave discrepancies give you the classic sound of some of the early monophonic synths. (And for musicians who use the Microcon as they would a bass synth such as Roland's TB-303, the imperfect octaves will probably not be a concern.) On the other hand, if you're playing a single line along with another instrument, the intonation could be a problem.

FILTER THIS

By using an insert cable in the output jack, you can process an external audio signal with the Microcon's filter: ring carries the audio input to the filter, and tip is the audio output. The ability to process external audio is a nice touch, making the unit that much more valu-

able. Plugging a mono miniplug into the gate jack mutes the Microcon's VCO so that you hear just the processed external signal at the output.

Another interesting feature of the Microcon is that the CV input can accept two control voltages from a miniplug insert cable. The tip of the stereo plug controls the pitch, and the ring controls the filter. Thus I was able to control the Microcon's pitch with a sequencer while changing the filter characteristics separately from a keyboard—and vice versa when I swapped the tip and ring plugs.

The Microcon's Envelope section presents some additional surprises. With the Release switch off and the Attack knob at 12 o'clock, the Trigger button triggers the envelope—just as you would imagine. However, once you turn the Attack knob to the left of 12 o'clock (so that the envelope is opening on its own), the Trigger button triggers the decay. In this case, if the Decay knob is set fully counterclockwise, pushing Trigger the sound. With the decay set longer, the envelope jumps to that setting and you immediately hear the decay. If you're running another synth through the Microcon's filter but the volume of the source synth is down, the Trigger button opens and closes the envelope, allowing you to hear the Microcon's LFO on its own. What's so interesting about all of this? If you want to perform with this unit (whether in the studio or on stage), knowing how the various parameters interact gives you a number of useful options for real-time control.

TECHNOSAURUS
Microcon analog synthesizer
$329

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

PROS: Sounds great and is small and easy to use. Has simplified signal flow. Filters can process external sound sources.
CONS: Comes with wall-wart power supply.

Microcon Specifications

Synthesis	additive
Polyphony	monophonic
Audio Outputs	(1) unbalanced ¼"
CV/Gate Inputs	(1) ⅛" TS; (1) stereo ⅛" TRS
Audio Oscillators/LFOs	1/1
Oscillator Waveforms	square; sawtooth
Envelope Generators	(1) AD or AR
VCF	(1) lowpass; 12 or 24 dB per octave
Signal-to-Noise Ratio	75 dB
Dimensions	8.11" (W) × 2" (H) × 4" (D)
Weight	1.21 lbs.

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WRH

MONSTROUS TONE

When it comes to rich analog sounds, the Microcon delivers. As both a bass and a lead instrument, this little synth has a lot to offer and is more versatile sonically than I originally expected.

Comparing the Microcon—and the Microcon II, for that matter—with other analog instruments is difficult because these units have a sound all their own. To my ears, the Microcon leans more toward an Oberheim SEM than anything else.

Played in combination with other synths, the Microcon blends nicely yet retains its own character in a mix. The ability to change the slope of the filter cutoff from 12 to 24 dB certainly helps in this regard. For example, I liked the way it sounded with my Sequential Circuits Pro-One.

If you're looking for another Roland TB-303 clone, you'll have to look elsewhere. But if you want a punchy portable synth, check out the Microcon.

16-STEP PROGRAM

The Cyclodon is a 16-step sequencer that is almost Zen-like in its stark simplicity. To begin with, it has two rows of eight continuously variable potentiometers, as well as an that LED indicates which row is being played. Four switches allow you to set the number of steps used in the sequence. The Run/Stop button controls the sequencer playback when the unit is in Auto mode; in

Manual mode it moves you through the sequence one step a time—perfect for tuning each step. The Clock switch puts the sequencer under the control of either the internal clock or an external signal. Under internal control, you can set the speed of the sequence using the Tempo knob.

FIG. 3: The Microcon II features a MIDI input on the left side of the unit, as well as a dual CV output from a stereo minijack on the right.

The Cyclodon sends control voltage and gate signals through minijacks on the unit's right side. The panel also includes a Roland-style 5-pin DIN sync jack so that you can run the Cyclodon from a drum machine, rhythm box, or MIDI-Sync24 converter.

ADDITION AND DIVISION

One interesting feature of the Cyclodon is the inclusion of four Step Number switches, marked +8, +4, +2, and +1. These switches allow you to create sequences of 1 to 16 steps, and you can change the number of steps while the sequencer is running. Want to do an additive sequencing piece in which you increase the number of steps in the sequence one at a time? With the Cyclodon, you can.

You can also control the number of steps that are played for each external pulse via the Clock Divider setting. It

can be set to play whole notes (one sequencer step for every four external beats), half notes (one step for every two beats), quarter notes (one step per beat), eighth notes (two steps per beat), and 16th notes (four steps per beat).

You set the clock division using the step-number switches—1 for whole notes, 2 for half notes, and so on. Unfortunately, you have to turn the external signal off before you can change the clock division. With a little practice, however, I was able to reset the clock division within a bar or two of a break, depending on the tempo.

QUIBBLES AND WISHES

After spending a few weeks working with the Cyclodon, I encountered a couple of minor problems that stuck in my craw. For starters, the Run/Stop button is too close to the Step Number switches; more than once I accidentally

MICROCON II

The Microcon II ($449) is the newest addition to Technosaurus's Small Monster family. The front-panel layout of the Microcon II is identical to the original Microcon's, but the control options are somewhat different.

To begin with, the Microcon II can be controlled by MIDI through the input jack on its left panel. However, you can control only performance aspects via MIDI; the current implementation doesn't let you save voice setups or patches. You can select the MIDI channel by using the dial next to the MIDI jack; the dial is marked in hexadecimal increments (0 through F) equivalent to MIDI channels 1 through 16.

Although the controller implementation is rather thin, I found many

uses for a small analog synth that I can play from any external MIDI controller. The parameters you can control via MIDI include Pitch Bend, Glide, Accent (on or off depending on the range of the Velocity), and the cutoff frequency of the VCF.

The Microcon II also gives you two CV outputs on one stereo minijack. With an insert cable I was able to play the original Microcon and my Oberheim SEM along with the Microcon II from a MIDI keyboard.

Speaking of stacking synthesizers, the VCO on the Microcon II has a range of a tritone (±6 semitones) above and below the 12 o'clock center position of the Tune knob. However, unlike the Microcon's, the

Microcon II's Tune knob is not center detented. Pitchwise, the center position yields something between A-440 and the G-sharp below it—slightly different from the manufacturer's specifications.

In all other matters the Microcon II behaves like the Microcon. Side by side, the two sound slightly different: the Microcon II is a tad fuller-sounding, due perhaps to the lower cutoff frequency of its filter. Because of its MIDI features, the Microcon II costs $120 more than the original Microcon, but I would recommend spending the extra money if you want the added control that MIDI offers.

Overall EM Rating (1 through 5): 4

Cyclodon Specifications

Steps	16
CV Output Range	0 to +5 VDC
Gate Output	0 or +5 VDC
Tempo Range	25 to 250 bpm
CV/Gate Outputs	(1) ⅛" TS; (1) stereo ⅛" TRS
Sync Input	(1) 5-pin DIN
Dimensions	8.11" (W) × 2" (H) × 4" (D)
Weight	1.21 lbs.

shut off a sequence while changing its length. Similarly, the Cyclodon's power jack isn't recessed far enough into the unit; the wall-wart plug protruded just enough that I could easily disconnect it if I brushed my arm against it. (The Microcon's power jack doesn't have this problem.)

Because of the Cyclodon's small footprint, features that are important to more advanced users of step sequencers had to be sacrificed. For example, a mute switch for each step—or at least a way to play the lower row of eight steps without the first—would be a nice addition. A transposition feature would also be welcome, as would a second set of CV and Gate outputs.

Of course, any added features would dramatically change the size and layout of the Cyclodon. Admittedly, I prefer Cyclodon's simplicity enough that I would forgo any additions (although

there just may be enough room for a second set of control outputs on the side panel).

MONSTERS ON THE HORIZON

Technosaurus has big plans ahead for its Small Monster series. The next device in the series is an analog multi-effects processor called the Effexon. The effects slated for this unit include overdrive, a VCO modulator, a 2-band parametric EQ, and a ring modulator that can use an internal or external control signal.

The Effexon is due for release by the middle of the year. Technosaurus also has plans for a module that will power a number of Small Monsters, eliminating the need for a cluster of wall warts and their attendant power strip.

A PERFECT PAIR

Both the Microcon and the Cyclodon are straightforward and simple to use, and either unit would be a significant addition to a live-performance or studio setup. DJs and dance musicians will appreciate the size, portability, and useful control configuration of these instruments. Studio synthesists will find that these Small Monsters can easily interface with older analog synths. The Cyclodon, for example, immediately fit into my studio setup. Because of its small footprint, it was especially convenient when used with my keyboard synths.

If you are new to analog gear, you'll appreciate the basic, no-frills aspect of the Microcon and Cyclodon as you familiarize yourself with the language of voltage control. Separately or together, the Microcon and Cyclodon are instruments well worth investigating.

TECHNOSAURUS
Cyclodon analog sequencer
$329

	1 2 3 4 5
FEATURES	■ ■ ■
EASE OF USE	■ ■ ■ ■ ■
DOCUMENTATION	■ ■ ■
VALUE	■ ■ ■ ■

PROS: It's small, lightweight, and easy to use. Includes sync input. Sequence length can be modified in real time.

CONS: Comes with wall-wart power supply. Power can come unplugged easily. Run/ Stop button is too close to step switches. Provides no way to mute steps. Can't run a lower row of voltages on its own.

SOUNDS LOGICAL

WAVEWARP 1.2 (WIN)

Real-time DSP for Windows.

By Thomas Wells

Now that DirectX and VST plug-ins offer so many ways to process digital audio, you might assume that most of the tools you need already exist. But what if you want a program that lets you build your own effects using dozens of different processes and functions in any type of configuration? Sounds Logical's *WaveWarp* is a stand-alone DSP toolkit with exactly that capability.

WaveWarp is an object-oriented audio-processing software environment for Windows. It provides an impressive collection of 250 signal-processing objects called *components* that connect via virtual patch cords to accomplish signal-processing tasks. Using this toolkit, you can create your own choruses, digital delays, equalizers, filters, flangers, reverbs, noise-reduction units, and more. All of these modules have real-time parameter control, but you won't find any live MIDI

input in the software's current version.

You can arrange components in arbitrarily complex networks consisting of parallel, series, feed-forward, and feedback connections. Components execute sample-by-sample in real time, using 32-bit floating-point arithmetic on any Pentium PC equipped with a Windows-compatible sound card. *Wave-Warp* 1.2 has full multichannel capability and supports multiple sound cards. The program can handle 8-, 16-, 20-, 24-, and 32-bit WAV files of any sample rate. It's even smart enough to optimize real-time performance by calibrating to your RAM and CPU configuration.

I tested *WaveWarp* on a Pentium II/400 MHz computer with 128 MB of RAM, and the program really smoked. To satisfy my curiosity, I also tried it on a 133 MHz machine with 64 MB of RAM (Sounds Logical's recommended minimum is a 166 MHz processor with 64 MB of RAM). Predictably, I got slower results, but they nevertheless fell well within an acceptable range of comfort. I began the review using version 1.1 (which had a number of minor yet bothersome bugs) and then moved on to the current release, version 1.2—a clean and robust piece of software.

REVVING THE ENGINE

Many electronic musicians are acquainted with the object-oriented sound-

WaveWarp
Minimum System Requirements
Pentium/166; 64 MB RAM; Windows 95/98/2000/NT 4.0; 16-bit sound card

construction environment typical of some software synthesizers and most sound design programs. Like other programs, *WaveWarp* 1.2 provides a work space (in its case, the Drawing Board) that wires together various components. The Drawing Board, on which I've designed a simple patch, as well as the program's other main work areas are displayed in **Fig. 1**.

Moving from top to bottom, you see the toolbar, the Component Library with its 250 objects and components grouped under 30 tab-style categories, the Component Library menu list for the selected category (Reverb), and the Drawing Board. Those who learn by hacking and cannibalizing will find that the library of 195 bundled Drawing Board examples is a terrific aid.

The patch in **Fig. 1** consists of two sound files, each processed through a ring modulator and reverb, then routed to a stereo digital-to-analog converter for playback. (Ring modulation occurs when the Sine Wave Generator multiplies the sample files, and the reverb is a function of the Moorer comb filter bank.) Building the patch was a snap: I simply selected each component from the appropriate area of the Component Category section and dragged it to the Drawing Board. After connecting the various outputs to the required inputs using *WaveWarp*'s virtual rubber-band patch cords, I had my design.

Double-clicking on a component opens its Properties window, where you'll find sliders, text-entry boxes, and other controls that adjust the component's values. The slider for Reverb time, which you can adjust during playback, appears on the right side of **Fig. 1**.

THE RIGHT CONNECTION

WaveWarp provides three types of virtual patch cables: thick black for stereo audio signals, thin black for mono audio signals, and thin red for control signals. To access the individual channels in a stereo audio signal, you use the Unzipper component to split the interleaved signal into two mono signals.
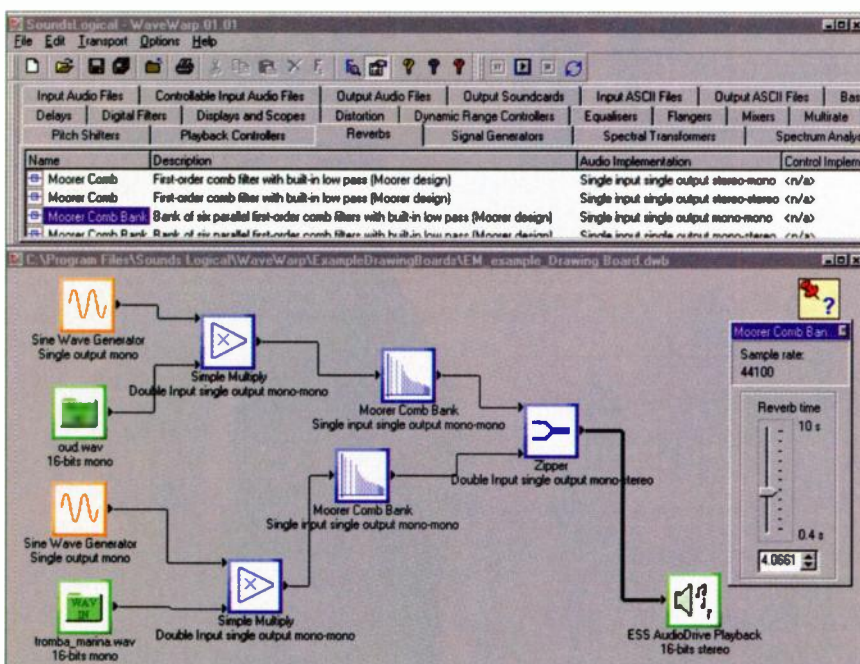
FIG. 1: The main window of Sounds Logical's *WaveWarp* shows (from top) the toolbar, component category tabs, and the selected category menu. The Drawing Board is the program's main work area and displays the individual modules that make up a patch.

- Brick Wall 1
- Brick Wall 2
- Brick Wall 3
- Brick Wall 4
- Controllable Brick Wall 1
- Controllable Brick Wall 2
- Controllable Brick Wall 3
- Controllable Brick Wall 4
- Controllable Spectral Shift 1
- Convolution
- Spectral 3-Peak Detector
- Spectral Averaging
- Spectral Compressor
- Spectral Expander
- Spectral Gate
- Spectral Peak Detector
- Spectral Shift 1
- Spectral Shift 2
- Spectral X-Synth 1
- Spectrum Shaper
- Tracking 3-Peak Detector
- Tracking Peak Detector

FIG. 2: The Spectral Transformers category includes 22 components for shaping your sound's spectrum.

In the patch shown in **Fig. 1**, I've "zipped" together the outputs from the two Mono-Output Multipliers before connecting them to the DAC (Audio-Driver) object; a mixer could perform that task as well.

Signal flow has certain restrictions. You can connect mono audio-signal outputs only to mono audio-signal inputs; likewise for stereo audio and control signals. However, various objects—for example, the Audio-to-Control component—can handle conversions. The components themselves are color-coded, and an *A* (for amplitude) or *F* (for frequency) marks control-signal components to indicate the expected range of the corresponding control I/O ports. Amplitude control signals typically fall in the numerical range of 0 to 1, and frequency control signals usually fall in the audio range of 50 to 20,000 Hz.

ABOUT THOSE COMPONENTS

WaveWarp offers an exhaustive array of signal-processing components, allowing you to build just about any effect imaginable. As previously mentioned, it includes 250 components grouped into 30 categories such as Signal Gen-

erators, Digital Filters, Mixers, Reverbs, Panners, Input Audio Files, Delays, Choruses, Distortion, Equalizers, and numerous others. (You'll find a complete list at the Sounds Logical Web site.) Some examples of the rich possibilities: the Spectral Transformers category features 22 components for manipulating a sound's spectrum directly (see **Fig. 2**); the Digital Filter category—a new benchmark for design quality—lists 20 types of filters; and the Signal Generator category offers 30 entries.

Purists might object to the number of (often overlapping) categories, but this system makes searching for components simple. My biggest concern with *WaveWarp* components is that you must double-click on them to access their control panels. These panels take up considerable desktop space, even when you set your graphics card to a low resolution. You'll encounter other minor inconsistencies as well—for example, the Input Audio Files category is actually two categories: one for sound files with playback-speed control, the other for sound files without control.

You may find the I/O scheme for components to be frustrating—some have interleaved stereo I/O and others have mono I/O. I eventually adapted to the system, but tiresome error messages such as the following marred the first few sessions: "You are attempting to connect a stereo audio signal to a mono audio input port. This is not allowed." In addition, you can't control an object by using an auxiliary device such as a graphics tablet or by tracking mouse coordinates, although Sounds Logical has announced plans for MIDI control in the next version of *WaveWarp*.

On the positive side, you can import wavetables, WAV files, and data for filter design from MATLAB—an integrated technical computing environment used by many DSP development engineers and researchers—into certain *WaveWarp* components. (Developers used MATLAB tools to create many of the DSP algorithms in today's commercial equipment.) This feature makes con-

siderable DSP design power available to *WaveWarp* users and sets the program apart from the competition. The Spectral Shaping object can be seen in action in **Fig. 3**; the Drawing Board with its components is on the left side of the screen, and the Spectral Shaping object's control window is on the right. The Spectral Shaping function imported from MATLAB appears on a graph in the Spectral Shaper component's control window. The graph is for display only—you can't use it to change the imported MATLAB function.

HEAR ALL ABOUT IT

Although *WaveWarp* currently lacks MIDI capabilities for real-time control of parameters, you can use it effectively to perform real-time manipulations of live audio inputs (subject to the usual latencies associated with audio I/O routed through Windows). Moreover, you'll have no trouble recording your sounds to disk for use in a sampler or digital audio program, or for burning onto a CD.

To create a WAV file of your design, you drag an Output WAV File component from the Output Audio Files category of the Component Library and drop it at the end of the signal chain. For slower PCs with low-end sound cards, Sounds Logical advises against running the Audio-Driver object that you'd normally use to hear a sound with the Output WAV component in parallel. The reason is that this may cause interruptions in playback (but not in the recorded sound file). So for performing effects in sync
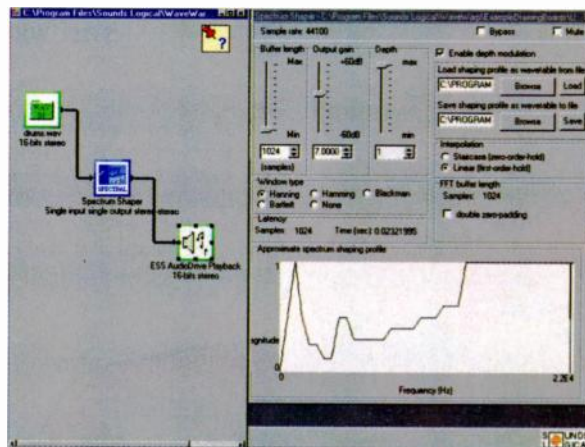
FIG. 3: You can incorporate data provided by MATLAB, a technical computing environment, into some *WaveWarp* components. In this figure, a spectral shaper uses data supplied by a MATLAB weighting function.

with, say, a sound file that you're processing, you'll need a fast PC.

To audition your freshly recorded material in *WaveWarp*, you simply open a new Drawing Board, click on the Input Audio Files tab, and scroll down through the list until you come to an object with the sound file's name (assuming you recorded it to the default directory). Drag that object onto the Drawing Board, connect it to an Audio-Driver object, click on Play in the toolbox transport (a Loop Play button is also available), and you're monitoring.

WaveWarp doesn't support playlists, but it does let you sequence multiple audio files with different start times. The procedure is cumbersome, requiring you to adjust the playback initiation times individually with the controllable playback function of each file. However, Sounds Logical has announced that the next version of *WaveWarp* will operate as a DirectX or VST plug-in that you can insert on a track-by-track basis into your sequencer application. It will let you use the sequencing and playlist features of your sequencer in conjunction with the powerful effects-design flexibility of *WaveWarp*.

You can input live audio to the program through a multichannel audio device for real-time processing. The sound quality of the modules is excellent, and the 32-bit floating-point processing avoids the clipping that some programs warn you about. With my Audiomedia III card as an intermediary between *WaveWarp*'s algorithms and my ear, I found the sonic results very pleasing.

GETTING HELP

WaveWarp's browser-based Help facility is complete, convenient, and, most important, accessible from inside the program during a session. For example, to get help with a given component, you click on it and press F1; your browser will display a thorough description of what the module does, its transfer function (in some cases), and a long list of related topics.

You'll find a second source of help in the toolbar at the top of the main window. Clicking on the yellow question mark opens the *WaveWarp* user guide; the blue one brings you to the Component Library Index; and the red one opens the Example Drawing Boards Index.

WaveWarp comes with a number of visualization components that help monitor and debug patches, including the well-implemented Oscilloscope and Spectral Display objects and the useful Audio Phase Scope and RMS Audio Display. The Oscilloscope appears in **Fig. 4**—two of them, in fact, are shown running with different time bases and displaying the same drum-track input.

In addition, the pushpin question-mark box on the Drawing Board (see **Fig. 1**) provides a notepad for user comments, and in the case of example Drawing Boards it contains details about the workings of a particular patch.

USER-FRIENDLY

In general, *WaveWarp* relies on a straightforward, simple design philosophy. Therefore, you may not have full access to all parameters of component objects. In some cases, this is understandable: you really can't control certain complex filters in real time. Some users might welcome the simplification, whereas others may find it constraining; it depends on how you're using *WaveWarp*.

According to Sounds Logical, *WaveWarp* is most suitable for three main groups of users. First are sound designers, musicians, audio-software designers, and hobbyists who want a high degree of control over the design of audio effects, with more flexibility than traditional plug-ins typically offer. The second group
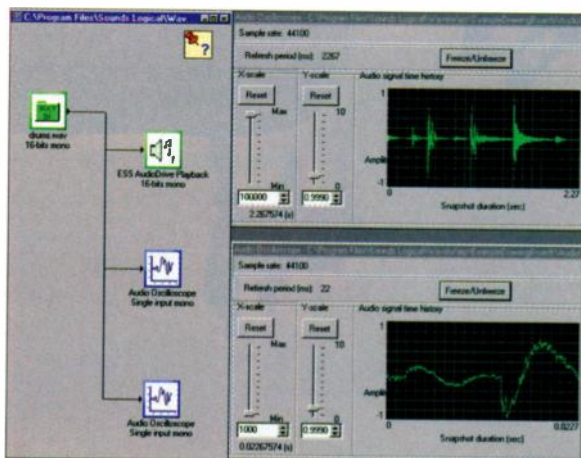
FIG. 4: Attach *WaveWarp*'s Oscilloscope to a signal chain for a view of the sound's content. Here, two oscilloscopes—each with a unique time base—track a drum loop.

includes professional engineers and researchers working in signal processing, acoustics, and spatial sound who wish to audition their designs in real time on a PC without any peripheral hardware or software. The last group consists of educators in the fields of electrical engineering, signal processing, acoustics, and audio technology who wish to illustrate key principles using live, real-time demonstrations featuring real-world audio signals.

WaveWarp is easy to use; even inexperienced users should have no problem navigating it. Intermediate and advanced users, however, may take issue with, say, its lack of versatility in sending messages that control the behavior of objects. Moreover, the absence of a developer's kit and the inability to create effects that can function as plug-ins in other software make *WaveWarp* less than ideal as a primary production tool. Yet its advanced DSP features, coupled with a development partner like MATLAB, allow *Wave-Warp* to offer tools that you won't find in similar programs on the market. The straightforward interface makes it very

intuitive for electronic musicians to use, and although you can't yet plug *Wave-Warp* designs into other host software or control the program through MIDI, where else can you get so much signal-processing power for the PC at such a reasonable price?

SUMMING IT UP
How well we work with a piece of software has a lot to do with its user interface. *WaveWarp*'s has a positive but somewhat constrained feeling, stemming from its inaccessible control parameters, lack of MIDI-input control, and inability to nest patches.

The program *is* fast: I built some large, redundant oscillator patches to probe its limits, and they worked at about 35 oscillators on my slower-than-recommended 133 MHz PC. On the 400 MHz machine, I tired of trying after reaching 100 oscillators, in part because of the onscreen clutter resulting from the lack of a nested-patch facility.

All things considered, with its speed and terrific variety of signal-generation and signal-processing options, *WaveWarp* is a great value for Windows-based musicians seeking a fast, complete effects-processing package.

Thomas Wells has been involved with audio and computer music for more than 25 years. He teaches at Ohio State University.
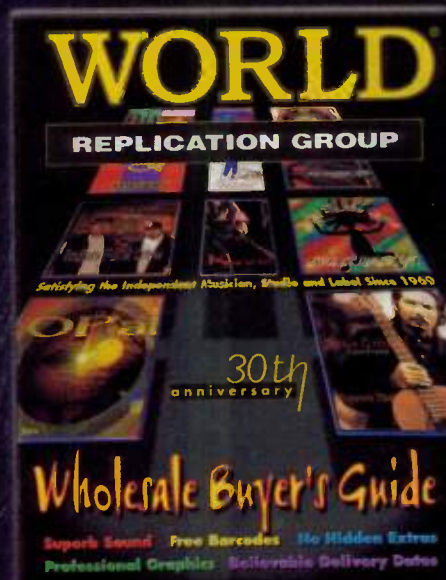
K O B L O

STUDIO9000 2.5 (MAC)

Rig your Mac with this bundle of virtual synths.

By Jeff Obee

This isn't a review of a Gary Burton CD, *A Streetcar Named Desire,* or the Greek alphabet. *Vibra, Stella,* and *Gamma* are Danish manufacturer Koblo's entries in the burgeoning software synthesizer market.

Koblo's Studio9000, a "rave in your Mac" bundle, consists of the *Gamma9000* drum machine, the *Stella9000* sample player, and the *Vibra9000, 6000,* and *1000* virtual analog synths (see the sidebar "Mini Vibras" for a description of the differences among the *Vibra* versions). *Gamma, Stella,* and *Vibra* share a common real-time synth engine named Tokyo.

MENU ITEMS

The Tokyo menu gives you access to the elements that all Studio9000 instruments share, most notably the audio and MIDI drivers. The supported audio drivers are Sound Manager, MAS, Direct I/O, DAE (Digidesign Audio Engine)/DigiSystem INIT, ASIO 2.0, and VST 2.0; the MIDI drivers are

OMS and FreeMIDI. If you use an Audiomedia II or III card, download Digidesign's 1.4.2 sound drivers. These system extensions allow Sound Manager to route your signal directly to the card instead of going through the Mac's built-in audio hardware.

Audio and MIDI drivers are assigned in Tokyo's Select Drivers window, where you can also access and edit your MIDI routing setup. With the Option menu's MIDI setup item, you choose the channel and instrument that will control your selected software synth, and determine whether the synth will transmit MIDI. For the Koblo synths to exchange MIDI data with your sequencer, all the devices must speak to each other via an interapplication communication bus. Interapplication audio routing is available with Steinberg's ReWire and VST 2.0, Mark of the Unicorn's MAS, and Digidesign's DirectConnect drivers.

Studio9000 programs are always referred to as *presets.* A default preset is included as a template for your own programs, which you save and access with the Presets menu. Presets are changed by using the + and – keys to move through the preset list, or by transmitting standard Program Change messages from your sequencer. However, when I sent Program Change messages from my sequencer (at 120 bpm), I noticed a lag of 1 to 1.5 beats.

The presets are grouped alphabetically by category in folders that ap-

pear as submenus in the Presets menu, rather than by the standard Program Change numbers (0 to 127). As a result, when you change a preset with a Program Change message, you must check the change by ear or find the selected preset in the Tokyo menu. One work-around is to create a new Presets menu folder and name it *a, b,* or something similar that will place it near the top of the list. Tokyo assigns lower Program Change numbers to patches appearing earlier in the list, so when you place your patches in the new folder, they will assume lower numbers. In this way you can ensure that your patch choices will be at the top of the list (see **Fig. 1**). A nice touch is the Project function, a macro that lets you save presets from all your Studio9000 synths as a group.
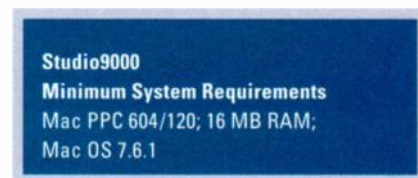
A SLEEK LOOK

All of the instruments have the sleek, modern appearance you expect in a turn-of-the-millennium Mac product. *Vibra* is green, *Stella* blue, and *Gamma* purple. Each synth has virtual dials that rotate when you move your mouse up or down.

Every parameter has an assigned Control Change (CC) number, which appears in the Global window when you click on a dial. You can capture the output of dial movements by recording it directly to your sequencer. Because there are so many dials, a list of the CC number assignments in the manual would be helpful.

The Studio9000 screens are designed for a 17-inch monitor and don't resize, so you have to scroll around the entire window to bring the hidden section into view—a headache. On the positive side, the instruments have only one screen each, so you won't need to perform this chore on a monitor cluttered with windows.

VIBRA FUN

Vibra9000 is a monophonic virtual analog synth that includes two oscillators (with FM and AM), eight filter types, three LFOs, three envelope generators, eight modulation sources and destinations,

FIG. 1: In the Presets menu, I've put my own patches in the folder named *a,* **gathering them at the top of the alphabetical list and thereby forcing the Tokyo synth engine to assign them the lowest Program Change numbers. I've called up "Atriditty" with Program Change message 0.**

FIG. 2: *Vibra9000*'s Filter window contains a menu of the various filter selections. I've selected the Saw Comb filter, with no highpass output, medium bandpass output, maximum lowpass output, and a bit of distortion.

and an arpeggiator. Its Global section has tuning, pan, and volume control dials. Dedicated buttons control functions such as triggering the synth without a keyboard and recording *Vibra*'s audio output to your hard drive. Other buttons call up a display of incoming MIDI data and activate Hold, Solo, Mute, and Panic. Three data windows show the currently selected parameter, its value, and the preset that it affects.

The oscillators use typical analog-style waveforms—sawtooth, square, triangle, sine, and noise, but no pulse. There's also no oscillator sync. I didn't really miss either feature because *Vibra*'s AM and FM capabilities offer seemingly endless sound-alteration possibilities.

The oscillator's Octave control spans a startling 15 octaves, so if you seek subsonic sounds, you've got 'em. When you're in the highest ranges, don't be surprised to find a pack of dogs outside your studio! Rounding out the oscillator section are Detune, Mix, Portamento, Bend, and a stereo on/off button.

BOSS TWEAK

Vibra's filters are downright boss (see **Fig. 2**). It has eight multistate filters with cutoff and resonance (2-, 4-, and 8-pole); double and quad (multiple parallel 12 dB filters); notch; and saw- and square-comb types (with multiple resonant peaks falling into a harmonic relationship or an odd-harmonic relationship, respectively). Your selected filter feeds highpass, bandpass, and lowpass outputs. One or more of these outputs must be turned up before you can hear the signal. The Spread parameter controls the balance of the final filter output. *Vibra* offers dis-

tortion, too, for some down-and-dirty patches.

The ADSR envelopes can be inverted, and the overall envelope amplitude is Velocity sensitive. The LFO waveforms are Ramp Up and Down, Triangle, Square, Sine, and Random. Each LFO has an Attack/Decay envelope and can synchronize to MIDI Clock (so you can sync the LFO to the tempo's whole notes, 16th notes, 32nd notes, and so on, including triplets and multiplied values). The Sharp parameter is a lowpass filter whose lower settings smooth the edges of the LFO waveform.

The LFOs can modulate nearly all of the parameters—envelopes, Aftertouch, Pitch Bend, and other data types such as Mod Wheel, Data Entry, Note On, and Velocity (yes, even modulators). Modulation doesn't noticeably kick in unless you set relatively high amounts of it, however. You may need to raise the source and destination levels for more intense effects.

Finally, *Vibra* has a single arpeggiator that can sync to MIDI Clock and run up, down, or up and down. The pattern, length, and Slide (portamento) are each controllable by note number, and you can alter the pattern's rhythm by minimum and maximum Velocity settings. If you desire multiple arpeggiators, you can open more than one copy of *Vibra9000*, or run *6000* or

1000 with it simultaneously if your CPU can handle the task.

MR. FRED SPEAKS

A number of fascinating presets included with *Vibra9000* show off its potential. Among these are "Japanese Cave," a shimmering, dipping effect; "Mr. Fred," a chic vocal formant patch; and "Narrow," which uses the arpeggiator with a bobbing rhythm and portamento.

The emulative sounds miss the mark, but if you're using a synth like this for emulative patches, we need to have a talk. This synth can traverse a lot of sonic terrain and is far more effective at producing imaginative, animated sound. *Vibra9000* is easy and fun to program, and it excels at arpeggiated textures, sound effects, and incidental enhancements. (I programmed some great lead and bass sounds, but latency in such usage could be an issue in real-time performance.) The synth sparkles when you dig into the modulation matrix; using an LFO to control FM or distortion, for instance, can shift a patch's texture dramatically.

STELLA!

Stella9000 is an 8-voice sample-playback instrument that supports up to 32-bit, 44.1 kHz samples in AIFF, SDII, and SampleCell formats (no WAV files at this juncture). The lack of sampling capability may be a drawback to some, but I found *Stella*'s simplicity refreshing. The interface resembles that of the *Vibra9000*, so I'll just explain the differences.

You assign samples for playback by clicking in the Sample field of *Stella*'s main screen (see **Fig. 3**). As with *Vibra*'s

MINI VIBRAS

Pared-down versions of *Vibra9000* also come in the Studio9000 package. *Vibra6000* has one oscillator with the same waveforms as in *9000*. It also includes octave, portamento, bend, and stereo on/off controls. The arpeggiator, filter, Global sections, and Control Change (CC) data control are exactly the same as in *9000*, and you'll also find Velocity sensing. But the versions differ in that the *Vibra6000* oscillator and filter each have only one (hard-wired) envelope, and the envelopes cannot be inverted. Another

difference is *Vibra6000*'s lack of modulation capabilities and LFOs.

Vibra1000 is available for free download from Koblo's Web site (www.koblo.com). Its lone oscillator (no stereo) has four waveforms (saw, square, triangle, and sine), and the filter consists of a lowpass with cutoff. Simple attack/release envelopes are dedicated to the filter and amplitude. The arpeggiator has all the parameters of its siblings. And *Vibra1000* lets you tweak CC data, too.

FIG. 3: *Stella*'s main screen is a compact display of sound and performance parameters.

oscillators, samples can have a 15-octave range. You can tune samples to semitones and cents and set them for the desired degree of Pitch Bend. The Off-set function changes the sample's starting point, enabling you to coax a lot of variety out of one sound source. The sample can also be looped or reversed.

Stella provides a basic Attack/Decay envelope, and Velocity can be routed to offset, pitch, volume, or pan.

Version 2.5 can read only SampleCell keymaps; if you have sounds mapped in that format, you're in luck; if not, only single samples can be imported, and you'll have to settle for a sound with a playable range that's less than optimal. This is an odd limitation, as the *Gamma9000* drum machine can freely distribute samples into keymaps (more on *Gamma* later).

Terrific on its own, *Stella* was a worthy companion to my Kurzweil sampler. The presets are excellent—"Blue Sky" and "Oceanic" are especially sumptuous pads—and the samples are readily available and easy to tweak. *Stella* also has several cool, ambient string-type pads.

Speaking of pads, creating new ones was a simple matter. I imported a pad into Emagic's *Logic Audio* from an audio CD, trimmed it, and saved it as an AIFF file. I then selected it in *Stella*'s Sample section and quickly had myself a killer backdrop texture.

I put the hard drive recording feature to the test here, too. I recorded a *Vibra*

FIG. 4: In *Gamma*'s Step Sequence section, you enter beats with the Hit buttons and activate accent, velocity, duration, and pitch with the Select buttons.

arpeggiator pattern to disk, then trimmed it using a waveform editor. I opened the new sample in *Stella* (with the loop button on), and it looped perfectly. I performed the same task with a sawtooth wave from *Vibra* and was able to play the sample polyphonically, bypassing *Vibra*'s monophonic limits.

GAMMA RAVES

Gamma9000 is designed to emulate older beatboxes such as the Roland TR-808. The upper half of the screen is devoted to *Gamma*'s seven tracks; the first six hold one sample each, and the seventh is available for a separate multisample keymap. You can use your own SDII or AIFF samples (or SampleCell keymaps), or select from the preset kits, which include the Roland TR-606, -707, -800, and -808; the Inn 9000; and the Unix Micro-Rhythm 12.

Gamma alters your sounds in ways not normally possible with a traditional drum machine. For example, you can offset, reverse, or loop a sample. A simple lowpass filter is adjustable by means of a tone control, and Velocity modulation can be assigned to tone, pitch, pan, volume, and offset. You get a simple envelope with attack and release, and there's a section for setting the sample's pitch and pitch modulation with the pitch-bend wheel. In the Mix section you can solo, mute, and trigger tracks, assign them to outputs (including output through the filter), and enter volume and pan settings.

A step sequencer containing a row of 16 step-entry buttons lets you create rhythmic patterns (see **Fig. 4**). Up to eight patterns can be recorded, looped, and assembled into songs. (You get four banks in which to store songs.)

To edit a pattern, you isolate a track by selecting its button, then click on the desired beats. You can easily accent a specific beat or alter its pitch, dura-

tion, or Velocity; manipulations such as reversing and randomizing can be performed on entire tracks. The Shuffle and Tempo functions provide additional creative control, and a tape-style transport controls playback when you're ready to let it fly.

More sound-shaping parameters are available in the Filter section. Here you'll find settings for cutoff, resonance, and distortion, along with buttons to activate highpass, bandpass, and lowpass. These features can give your patterns considerable character.

One fun way of using *Gamma* is to assign nontraditional percussive timbres, such as truncated vocals, animal sounds, and industrial samples, to the tracks. The step-entry interface provides a different and inspirational perspective on rhythms, and using weird or twisted sounds is simply a gas.

SKIP, HOP, AND WOBBLE

To test *Gamma*'s performance, I closed my other applications, cleared existing patterns, and clicked in four bars of 16th-note hi-hat attacks. Listening carefully, I noticed slight lags in what should have been a very steady rhythm. *Gamma* fared better with intermittent hi-hat rhythms and more sparsely played snares and kicks.

I then used *Gamma* as a sound source with *Logic Audio* as the sequencer (bypassing *Gamma*'s internal sequencer). The timing was much better, so consider this option if you encounter the same performance problems.

THE KITCHEN SYNC

One element that makes Studio9000 especially appealing is its ability to sync to MIDI Clock. *Gamma* has this feature, as do the LFOs and arpeggiator in *Vibra* and *Stella*. This enables all the Koblo instruments to work as a synchronized team in a sequencing environment.

I used a Power Mac 7100 with a Newer Technology G3/300 MHz CPU accelerator card and OS 8.1 for this review. Those of you who own a stock Mac G3 or G4 will attain superior performance with their faster CPUs and system buses.

I first tested the *Vibra* and *Stella* arpeggiator sync by running them with *Logic Audio* in a composition. I used both (at different points in the piece), with 16th-note arpeggiator patterns at 166 bpm. They performed beautifully; they also responded well when I modulated parameters on the fly

using the Keyfax PhatBoy controller.

The ultimate test was running all five Koblo synths at once with *Logic Audio: Gamma* played a fast techno rhythm, *Stella* a pad, *Vibra 9000* an arpeggiator pattern, *Vibra6000* a bass line, and *Vibra1000* another arpeggiator pattern. In this context Studio9000 had difficulty with timing. The sequencer stuttered and everything dragged.

Undoubtedly, the problems were due to my Mac's limitations (even though my system runs faster than the recommended minimum speed). Running

Logic Audio, OMS, and Tokyo at the same time consumed most of my 80 MB of RAM. Tokyo alone uses 40 MB, so Koblo's recommendation to use 128 MB should be well heeded. The keys to optimum software-synth performance are a fast CPU, a fast system bus, more RAM, and high-quality audio I/O.

CODA

Let me make my stance perfectly clear: these synths are very cool. Studio9000 is relatively inexpensive; it resides in your computer (freeing up studio space); and *Gamma, Stella,* and *Vibra* all have a hip sound. Digidesign recently picked up the Koblo line for distribution, so you can integrate Studio9000 into Pro Tools systems, and the bundle is compatible with Digidesign audio drivers.

Studio9000 won't replace my hardware synths, and I doubt that I'll use it as the sole source of sounds for a whole tune. But the programs are certainly useful as add-ons to my main rig, and their simple yet powerful architecture complements my musical style. Give the demos a try and see how these synths fit your style.

Jeff Obee is a bassist and synthesist whose CD Obee Sings Mary Poppins Whilst Wearing a Tutu *has been turned down by several record companies. You can reach him at obeej@dsp.com.*

QUICK PICKS

SAMPLEHEADS
Rhythm of the Americas: Latin Drums, Volume 1
By Alex Artaud

● ●

Anyone familiar with Sampleheads' sample CDs knows that they're meticulously recorded. The recently released *Rhythm of the Americas: Latin Drums,* volume 1,
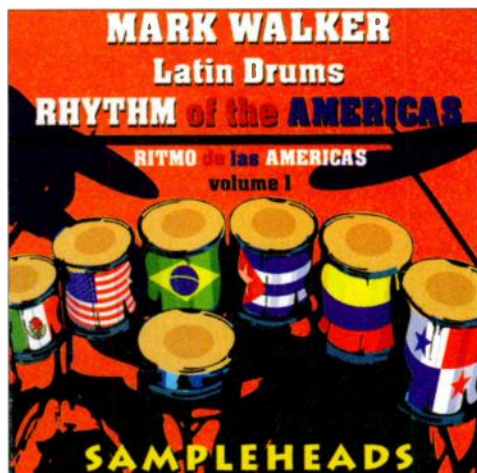
Sampleheads' *Rhythm of the Americas: Latin Drums,* volume 1, sample CD offers a comprehensive selection of authentic Latin drum grooves at a variety of tempos, as well as a useful assortment of drum hits.

(audio CD, $99.95; CD-ROM, $149.95) lives up to that reputation, offering 72 minutes of crisp, consistent performances of popular Cuban and Brazilian rhythms played by drummer Mark Walker. These solid fills, grooves, and hits will lend a spicy flavor to your sampling.

Walker brings a relaxed, authoritative vibe to this effort, perhaps as a result of his work over the past decade with top-notch Latin ensembles including those of Paquito D'Rivera, Michel Camilo, and Claudio Roditi. His confidence and energy are evident throughout the CD.

¡Saaaaabor!
Latin Drums presents 99 concisely organized tracks, each containing multiple arrangements of a given rhythmic pattern.

Samples on this disc range in tempo from 84 to 148 bpm.

The first five track groupings consist of Cuban rhythms—Songo, Funky Songo, Mambo, Cascara, and bell patterns. The Songo and Funky Songo grooves showcase the fusion of folk rhythms (found in traditional styles like rumba) with patterns inspired by the rock and funk that influenced Cuban musicians in the 1970s.

In the Mambo tracks, parts that are typically played on bongos, congas, and timbales are transferred seamlessly to the drum set. The cascara patterns are supporting rhythms that the *timbalero* plays on the side of a timbale drum during quieter passages. Walker plays cascara patterns on the side of the floor tom and on hi-hat. Variations on common bell patterns round out the first section. Walker's playing is lively and propulsive throughout.

The next three sections cover Brazilian rhythms, including Batucada, Samba, and funky Partido Alto. The Batucada examples evoke Carnaval's vibe with its large percussion ensembles, while the Samba selections have a cool bossa nova sensibility. The Samba section also includes a selection of useful snare and tom fills.

The Partido Alto rhythms sound the least Brazilian but actually come from variations on Samba rhythms. *Partido alto* (high part) refers to a rhythm practiced by a select group of players in the Escola de Samba, the Brazilian organization that puts on samba parades for Carnaval. The spirit here is loose and adventurous.

Cream of the Crop
The CD concludes with a comprehensive "hits" section composed of kick, snare, tom, ride, hi-hat, crash, rim-shot, cross-stick, timbale, and cowbell samples. Flams, ruffs, and rolls are provided for added flexibility. This last section could have been a throwaway; instead, it's one of the CD's highlights, with a great assortment of sounds and dynamics. If you want to construct your own kit on a sampler, this disc will serve you well.

Latin Drums was recorded through a Neve Capricorn digital board into a Pro Tools/24 system, and the production values are fantastic. The documentation is excellent as well, providing detailed information such as a description of the clave's

importance and implied presence in all the Cuban rhythms. Each rhythm group is briefly introduced, and a thorough track listing covers all the patterns played and their individual times.

I reviewed the audio-CD version, but *Latin Drums* is also available as a CD-ROM in Akai, E-mu, Roland, WAV, SampleCell, GigaSampler, and SoundFont formats. Unlike some sample CDs, this collection includes no MIDI files, but you won't miss them. The great playing, wide variety of material, and pristine sound quality make this disc a winner.

Overall EM Rating (1 through 5): 5

LUCID
ADA8824
By Zack Price

● ●

The Lucid ADA8824 ($3,295) is a 2U, 24-bit A/D/A front end for ADAT-compatible audio cards and digital audio workstations. It transfers up to eight channels of audio simultaneously and operates at a maximum sampling rate of 48 kHz. The ADA8824 Sonic, another member of the Lucid family, is available for use with Sonic Solutions' SonicStudio. I reviewed the ADAT version, but the units offer similar features.

Up Front and Around Back
The ADA8824's front panel has a 15-segment LED meter for each of its eight audio channels, as well as a conveniently located power switch. The Sync window to the left of the meters displays the synchronization source. Although five sync options are available (44.1 kHz internal, 48 kHz internal, word clock, AES/EBU digital input, and ADAT sync), the front panel shows only the first four. The ADA8824 highlights the active options; it remains unlit when the ADAT-sync option is activated.

The Metering window to the right of the channel LEDs shows the audio sources being metered (analog or digital input and analog or digital output). DIP

The Lucid ADA8824 is a high-quality 24-bit converter for use with ADAT-compatible devices or Sonic Solutions' SonicStation. It provides eight channels of I/O at a sampling rate of up to 48 kHz.

CD-RELIABLE

switches located on the unit's rear panel change both the sync and metering options. You can adjust settings via MIDI with SysEx messages or through the supplied Macintosh or Windows controller software. Windows 95 or 98 users should control the ADA8824 with a vacant MIDI port or a dedicated interface, because Windows doesn't allow MIDI devices to be shared simultaneously by different programs. (Mac users running OMS don't have this limitation.)

The rear panel has eight female XLR jacks for analog input, as well as eight male XLR jacks for analog output. It also has four female and four male XLR jacks for AES/EBU digital input and output, respectively. Any of the first three AES/EBU inputs can be selected as external-sync inputs. The ADA8824 also includes S/PDIF I/O on female RCA connectors, which can accept an external sync signal.

Word-clock I/O uses BNC connectors; its output is synchronized to the AES/EBU outputs using one of two methods. If the AES/EBU outputs are fed from the analog inputs, the word-clock output gets its signal from the internal clock; if the AES/EBU outputs are fed from the ADAT optical input, the word clock derives its signal from the ADAT data stream.

The ADA8824 also features two female DB-9 connectors for sending and receiving ADAT-sync signals. Although these signals have three components (machine control, ADAT word clock, and time code), the ADA8824 doesn't use the ADAT word-clock signal. Moreover, the ADA8824 needs to be first in any chain of ADAT devices, severely limiting the use of its ADAT-sync option.

Sound Off
The ADA8824 sounds terrific. However, with its substantial price tag, it's obviously not for everyone. This unit may be more product than you need in a personal studio. Besides, using it for overdubbing is difficult unless your ADAT-compatible card also has word-clock capability.

Nonetheless, owners of post-production facilities or high-end personal studios will find the ADA8824 to be a great-sounding device with numerous digital audio routing possibilities. Its controller software simplifies configuring these features on the fly, and its word-clock I/O enables it to work easily with house sync. If you're looking for a high-quality 8-channel, ADAT-compatible A/D/A unit, check out the Lucid ADA8824.

Overall EM Rating (1 through 5): 4.5
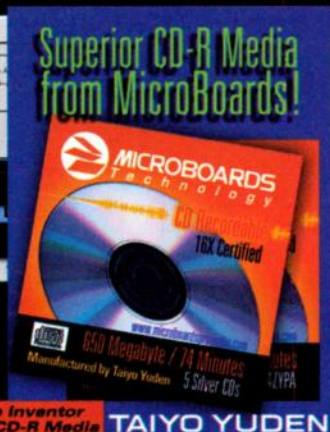
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multisampled pianos—Pop Piano and Brilliant Classical. Both are available with and without sustain pedal. The Akai and E-mu versions of these pianos use approximately 60 to 70 MB of RAM; the GigaSampler version uses around 200 MB.

Pop Piano is bright and edgy, but transitions between the hard and soft samples

Besides standard samples of the instrument, Best Service's *Total Piano* features unusual sounds such as prepared pianos, tack pianos, and piano effects.

of this Velocity-switched program sound awkward to me. It works better within a track than as a solo piano sound, but I prefer Brilliant Classical for all styles—including pop music.

I used the Brilliant Classical piano extensively in several recording projects. It sounds more authentic to my ears than many piano programs in the higher-priced libraries that I own. And despite the name, it's very versatile.

Both Pop Piano and Brilliant Classical suffer from a slight high-end fuzziness, possibly because these pianos—including the pedaled and unpedaled samples—are processed and sampled at 22 kHz to reduce memory consumption; the rest of the library is sampled at 44.1 kHz. Brilliant Classical gives you the option of replacing the 22 kHz fortissimo samples with 44.1 kHz versions (the program is Velocity-switched between mezzopiano and fortissimo samples), although the sonic difference isn't dramatic.

Not the Norm
Where *Total Piano* really shines is in its collection of unorthodox sounds—all made with a real piano. Several programs, found mostly on disc 2, center on unconventional playing techniques such as muted, Harpsichord (a piano played with a metal plectrum), Cimbalon (struck with wooden

mallets), and Circular Saw (stimulated with a motor-driven cogwheel). Hyper Piano and Magic Piano feature unique and ethereal timbres but were created without the use of outboard effects.

Several prepared-piano programs are available: Thai Piano (metal screws between the strings), Rubbery Piano (pieces of rubber between the strings), Honky Tonk, and my favorite, Thumb Tack Piano (thumbtacks stuck into the hammers).

The Piano Effects category provides an incredible 170 MB of outstanding special effects and atmospheres. These evocative clusters, glissandos, kicks, knocks, sweeps, hits, and "drum" sounds are destined to enhance many compositions and serve as creative springboards for further sound design.

Total Piano's best feature is how well all the various timbres of the library work together. In this collection, Best Service succeeds in maximizing the piano's potential. *Total Piano* is a winner.

Overall EM Rating (1 through 5): 3.5

STEINBERG
Sounds & Cycles (Mac/Win)
By Erik Hawkins

● ●

If you use Steinberg *Cubase VST* and are not acquainted with the ReCycle Export (REX) file format, then you're missing out on a world of powerful production sounds. REX files are sampled loops, cre-

ated in Steinberg's *ReCycle* loop-editing software, that have been beat-mapped; this allows them to be played back at a wide range of tempos with no change in pitch. (For more information on REX files, see the *ReCycle* review in the December 1999 issue of **EM**.) Currently, *Cubase VST* is the only application that reads the REX file format.

Steinberg has begun commissioning and distributing REX loop libraries. Sounds & Cycles, by Sounds Good, is one such library. It consists of six boxed sets: *Chemical Big Beats, Cold Fusion Technologic, Drum'N'Bass X-Citer, The Electro Age, Hip Hop-Beats and Treats,* and *On a Latin Tip.* Each sells for $59. This review covers *Cold Fusion Technologic* and *On a Latin Tip,* and gives some general notes on the library as a whole.

The Big Package
All of the boxed sets in Sounds & Cycles are packaged and organized similarly. Each set is made up of four CDs. Discs 1 and 2 hold AIFF, WAV, and Red Book audio files. If you listen to the audio files, be sure to start playback from track 2 because track 1 contains data that will generate a nasty sound in your speakers. Disc 3 has all the REX files, and disc 4 is a Steinberg product demo CD-ROM.

The loops in *On a Latin Tip* are organized first by instrument and then by tempo. For example, the Bass folder holds five subfolders, each labeled with tempos ranging from 70 to 120 bpm in increments of 10 and containing bass lines with the corresponding tempo. *Cold Fusion Technologic*

The Sounds & Cycles library, programmed by Sounds Good and distributed by Steinberg, consists of six boxed sets of looped REX-format files in addition to AIFF, WAV, and Red Book audio versions of the loops.

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is organized first by tempo and then by instrument. Folders are labeled by tempo, from 120 to 150 bpm, in increments of 10. Within each of these folders are subfolders of bass, drum, and melody loops. As with time-stretching and compression techniques, you'll find that REX files are limited as to how much you can change their tempos. Loops with lots of sustained sounds aren't as flexible as loops with mostly staccato sounds, but the results depend largely on the sample itself. In any event, having the files organized by bpm is very convenient.

Whereas the AIFF, WAV, and Red Book audio files are stereo, the REX files are mono, which is standard for this format. The drum and instrument sounds you hear in the loops (such as kick, snare, bass, and synth arpeggios) are separated out as discrete AIFF, WAV, and audio files. These individual samples are great for augmenting your REX compositions because they allow you to program your own fills and breakdowns using the identical sounds heard in the loops themselves.

Key signatures for the tuned-instrument samples (for example, piano, guitar, and synth bass) are documented in the CD booklet. Often, loop libraries don't include this important information, so Steinberg's thoroughness is much appreciated. Sample categories are listed by audio-track number—for instance, tracks 1 and 2, Bass Loops. The samples themselves are not individually named, which makes it hard to tell just how many sounds you get in all; I'd estimate that there are more than 200 sounds on each disc. Everything is in 4/4 time.

In the Groove

If you want to add a little spice to your productions, *On a Latin Tip* offers some choice cuts. You can choose from swinging conga patterns, syncopated agogo riffs, cool sax lines, tropical guitar, melodic vibe samples, and smooth bass loops. The Latin jazz piano sounds are particularly nice. The Rhodes piano samples are cool, but unfortunately this category offers fewer files. I found several solid drum and percussion loops that are perfect for building a Latin-flavored percussion bed. The guiro samples are especially distinctive and useful.

Cold Fusion Technologic has some hot beats. The drum and percussion loops are loaded with deep electronic and techno elements. I used them in several of my electronica compositions and moodier dance pieces. I had a difficult time working with the tuned synth loops because of their choppy rhythms and dissonant overtones. However, you can combine a few amusing melody lines with generous doses of delay to create original polyrhythmic, ambient backdrops. Not many bass loops grabbed my attention—most were too busy with too many octave jumps to fit into my productions—but that's just me. If you're into dark minimalism and an underground sound, then these bass loops might be just your cup of tea.

The Big Bang

The Sounds & Cycles CDs offer a great bang for the buck considering all the loops, drum hits, riff samples, and audio formats you get for just over half a C-note. If you're a dedicated *Cubase VST* user, I recommend checking out this library. It gives you all the advantages of REX files without the hassle of having to make them yourself in *ReCycle*. And even if you don't use *Cubase VST,* the AIFF, WAV, and Red Book audio files could be useful to any loop fanatic. I look forward to trying out other Sounds & Cycles boxed sets. ●

Overall EM Rating (1 through 5): 4

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DIGITAL MULTI-TRACK RECORDERS

TASCAM

MX-2424 24-Bit 24-Track Hard Disk Recorder

Co-designed by TASCAM and TimeLine Inc., the MX-2424 is an affordable 24-bit, 24-track hard disk recorder that also has the editing power of a digital audio workstation. A 9GB internal hard drive comes standard as well as a SCSI Wide port that supports external LVD (Low Voltage Drives) hard drives from up to 40 feet away. An optional analog and several digital I/O cards are available so the MX-2424 can be configured to suit your work environment. SMPTE synchronization, Word Clock, MIDI Time Code and MIDI Machine Control are all built in for seamless integration into any studio.

- Records 24 tracks of 24-bit audio at 44.1 or 48 kHz, or 12 tracks at 88.2 or 96 kHz. Up to 24 tracks can be recorded simultaneously using any combination of digital and analog I/O.
- Supplied 9GB internal drive allows 45 minutes of audio across all 24 tracks
- Wide SCSI port on the back panel allows you to add multiple drives. A front 5-1/2" bay available for installing an additional drive, or an approved DVD-RAM drive for back-up.
- ViewNet MX, a Java-based software suite for Mac and PC offers DAW style editing of audio regions, dedicated system set-up screens that make set-up quicker and easier and track load screens that make virtual track management a snap. Connects to a computer via a standard Ethernet line.
- Can record to Mac (SDII) or PC (.WAV) formatted drives, allowing later export to the computer. The Open TL format allows compatible software to recognize virtual tracks without have to load, reposition and trim each digital file.

Transport Controls-
- Jog/scrub wheel
- MIDI In, Out, and Thru ports are built-in for MIDI Machine Control.

Editing-
- Built-in editing capabilities include cut, copy, paste, split and ripple or overwrite
- 100 levels of undo
- Supports destructive loop recording and non-destructive loop recording which continuously records new takes without erasing the previous version.

Build-In Synchronization-
- TBUS protocol can sample accurately lock 32 machines together for 384 tracks at 96kHz, or 768 tracks at 48kHz.
- Can generate or chase SMPTE timecode or MIDI Time Code.
- Word Clock In, Out, and Thru ports

I/O Options-
- Optional analog and digital cards all provide 24 channels of I/O. There is one slot for analog and one for digital.
- IF-TD24- T/DIF module
- IF-AD24- ADAT Lightpipe module
- IF-AE24- AES/EBU module
- IF- AN24- A-D, D-A I/O module with DB-25 connectors

Software Updates-
- System updates are made available through a front panel Smart Card slot or via computer directly from the TASCAM web site

DA-78HR Modular Digital Multitrack

The DA-78HR is the first true 24-bit tape-based 8-track modular digital multitrack recorder. Based on the DTRS (Digital Tape Recording System) it provides up to 108 minutes of pristine 24-bit or 16-bit digital audio on a single 120 Hi-8 video tape. Designed for project and commercial recording studios as well as video post and field production, the DA-78HR offers a host of standard features including built-in SMPTE Time Code Reader/Generator, MIDI Time Code synchronization and a digital mixer with pan and level controls. A coaxial S/PDIF digital I/O allows pre-mixed digital bouncing within a single unit, or externally to another recorder or even a DAT or CD recorder. Up to 16 DTRS machines can be synchronized together for simultaneous, sample accurate control of 128 tracks of digital audio.

Features-
- Selectable 16 bit or 24 bit High Resolution audio
- 24 bit A/D and D/A converters
- >104dB Dynamic range
- 20Hz - 20kHz frequency response ± 5dB
- 1 hr. 48 min. recording time on a single 120 tape
- On-Board SMPTE synchronizer - chase or generate timecode
- On-Board support for MIDI Machine Control

- Internal digital mixer with level and pan for internal bouncing, or for quick mixes
- Track slip from -200 to +7200 samples
- Expandable up to 128 tracks (16 machines)
- Word Sync In/Out/Thru
- Analog output on DB25 balanced or RCA unbalanced
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Features–
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- 6.2GB IDE hard drive
- Editing functions include move, divide, combine or delete audio tracks, add or drop any index or sub-index, and create track fade in or fade out
- Coaxial SP/DIF or AES/EBU digital input plus optical S/PDIF I/O
- XLR balanced and RCA Line inputs and outputs
- Automatic sample rate conversion from 32 and 48kHz

- Automatic CD Format Detection feature and user friendly interface provide one touch button operation
- Front panel trim pot and LCD display provide accurate input signal and time lapse metering
- SCMS (Serial Copy Management System) is supported, regardless of the source disc copy protection status
- StartREC Models Include: ST2000 (2) 8x writers, ST3000 (3) 8x writers and ST4000 (4) 8x writers

DIGITAL MIXERS

Roland

VM Basic 72 Digital Mixing System

The all digital Roland V-Mixing System, when fully expanded, is capable of mixing up to 94 channels with 16 stereo (32 mono) onboard multi-effects including COSM Speaker Modeling. Utilizing a separate-component design, comprised of the VM-C7200 console and VM-720C rackmount processor, allows the V-Mixing System to be configured to suit your needs. Navigation is made easy via a user friendly interface. FlexBus and EZ routing capabilities as well as a large informative LCD and ultra-fast short cut keys.

Features-
- 94 channels of digital automated mixing (fully expanded)
- Up to 48 channels of ADAT/Tascam T-DIF digital audio I/O with optional expansion boards and interfaces
- Separate console/processor design
- Quiet motorized faders, transport controls, total recall of all parameters including input gain, onboard mixer dynamic automation and scene memory
- 24 fader groups, dual-channel delays, 4-band parametric channel EQ + channel HPF
- FlexBus and "virtual patchbay" for unparalleled routing flexibility

Options-
- VS8F-2 Effects Expansion Board -- Provides 2 stereo effects processors including COSM Speaker Modeling. Up to 3 additional boards can be user-installed into the VM-7200 processor, for 9 stereo or 16 mono effects per processor.
- VM-24E I/O Expansion Board – Offers 3 R-Bus I/Os on a single board. Each R-Bus I/O provides 8-in/8-out 24-bit digital I/O, totalling 24 I/O per expansion board.

- Up to 16 stereo (or 32 mono) multi-effects processors using optional VS8F-2 Effects Expansion Boards (2 stereo effects processors standard)
- COSM Speaker Modeling and Mic Simulation technology
- 5.1 Surround mixing capabilities
- EZ Routing allows mixer settings to be saved as templates
- Realtime Spectrum Analyzer checks room acoustics in conjunction with noise generator and oscillator
- Digital cables between processor and mixer can be up to 100 meters long- ideal for live sound reinforcement.

- **DIF-AT Interface Box for ADAT/Tascam** -- Converts signals between R-Bus (VM-24E expansion board required) and ADAT/Tascam T-DIF. Handles 8-in/8-out digital audio. 1/3 rackmount size.
- **VM-24C Cascade Kit** -- Connects two VM-Series processor units. Using two VM-7200 processors cascaded and fully expanded with R-Bus I/O, 94 channels of audio processing are available.

EFFECTS PROCESSING

lexicon

MPX-500 24-Bit Dual Channel Effects Processor

The MPX 500 is a true stereo 24-bit dual-channel processor and like the MPX100 is powered by Lexicon's proprietary Lexichip and offers dual-channel processing. However, the MPX 500 offers even greater control over effects parameters, has digital inputs and outputs as well as a large graphics display.

- 240 presets with classic, true stereo reverb programs as well as Tremolo, Rotary, Chorus, Flange, Pitch, Detune, 5.5 second Delay and Echo
- Balanced analog and S/PDIF digital I/O

- 4 dedicated front panel knobs allow adjustment of effect parameters. Easy Learn mode allows MIDI patching of front panel controls.
- Tempo-controlled delays lock to Tap or MIDI clock

t.c. electronic

M-One Dual Effects Processor

The M-One allows two reverbs or other effects to be run simultaneously, without compromising sound quality. The intuitive yet sophisticated interface gives you instant control of all vital parameters and allows you to create awesome effects programs quickly and easily.

- 20 incredible TC effects including, Reverb, Chorus, Tremolo, Pitch, Delay and Dynamics
- Analog-style user interface
- 100 Factory/100 User presets

- Dual-Engine design
- 24 bit A/D-D/A converters
- S/PDIF digital I/O, 44.1-48kHz
- Balanced 1/4" Jacks - Dual I/O
- 24 bit internal processing

D-Two Multitap Rhythm Delay

Based on the Classic TC2290 Delay, the D-Two is the first unit that allows rhythm patterns to be tapped in directly or quantized to a specific tempo and subdivision.

- Multitap Rhythm Delay
- Absolute Repeat Control
- Up to 10 seconds of Delay
- 50 Factory/100 User presets

- 24 bit A/D-D/A converters
- S/PDIF digital I/O, 44.1-48kHz
- Balanced 1/4" Jacks - Dual I/O
- 24 bit internal processing

MICROPHONES

B&H is proud to announce that we are now dealers for RØDE microphones and all other Event products.

RØDE
NT-2 Condenser Mic

The RØDE NT2 is a large diaphragm true condenser studio mic that features both cardioid and omnidirectional polar patterns. The NT-2 offers superb sonic detail with a vintage flavor for vocal and instrument miking. Like all RØDE mics the NT-2 is hand-assembled in Australia and is available at a breakthrough price.

Features
- Dual pressure gradient transducer
- Large diaphragm (1") capsule with gold-sputtered membranes
- Low noise, transformerless circuitry
- Omni and cardioid polar patterns
- High pass filter switch
- -10dB pad switch
- 20Hz-20kHz frequency response
- 135dB Max SPL
- Gold plated output connector
- Gold plated internal head pins
- Shockmount, Flight Case, and Pop Filter included

SHURE
KSM-32SL
Cardioid Condenser Mic

The reviews are raving about Shure's new "classic" microphone. The KSM32 features Class A, transformerless preamplifier circuitry, low self-noise and increased dynamic range, all necessary for critical studio recording. It has a 15 dB attenuation switch for handling high SPLs, making it suitable for a variety of sound sources including vocals, acoustic instruments, ensembles and overhead miking of drums and percussion. For studios, the KSM32/SL has a light champagne finish and includes an aluminum carrying case, shock and swivel mounts and a velvet pouch. For live applications, the KSM32/CG has a charcoal grey finish and includes a swivel mount and padded zipper bag.

- Frequency response 20Hz - 20kHz

AKG
C4000B Electret Condenser Mic

This new mic from AKG is a multi polar pattern condenser microphone using a unique electret dual large diaphragm transducer. It is based on the AKG Solid Tube design, except that the tube has been replaced by a transistorized impedance converter/ preamp. The transformerless output stage offers the C4000B exceptional low frequency response.

FEATURES-
- Electret Dual Large Diaphragm Transducer (1st of its kind)
- Cardioid, hypercardioid & omnidirectional polar patterns
- High Sensitivity
- Extremely low self-noise
- Bass cut filter & Pad switches
- Requires 12, 24 or 48 V phantom power
- Includes H-100 shockmount and wind/pop screen
- Frequency response 20Hz to 20kHz

GT
AM-61 Cardioid Tube

The GT Electronics AM61 offers classic tube performance in a fixed cardioid, large diaphragm condenser mic. An outstanding addition to any project studio or large commercial recording facility seeking rich, warm tube sounds and unsurpassed value.

- Groove Tubes military-spec GT5840M vacuum tube preamplifier
- Large-diameter, super-thin 3 micron gold evaporated Mylar diaphragm
- Fixed cardioid polar pattern response
- Switchable -10dB attenuation pad and 80Hz low frequency roll-off filter
- Includes hard-shell case, shock mount, hard mount, 6-pin cable and external power supply
- Frequency response 20Hz - 20kHz

- ALSO AVAILABLE AM-62 multipattern tube condenser mic

audio-technica.
AT4047SV
Cardioid Condenser Mic

The AT4047 is the latest 40 Series large diaphragm condenser mic from Audio Technica. It has the low self noise, wide dynamic range and high sound pressure level capacity demanded by recording studios and sound reinforcement professionals.

- Side address cardioid condenser microphone for professional recording and critical applications in broadcast and live sound
- Low self noise, wide dynamic range and high SPL
- Switchable 80Hz Hi Pass Filter and 10dB pad
- Includes AT8449/SV shockmount
- **Also Includes a limited edition tweed flight case while supplies last!**

STUDIO MONITORS

VERGENCE A-20
Studio Reference Monitor System

Incorporating a pair of 2-way, acoustic suspension monitors and external, system-specific 250 watt per side control amplifier, the A-20 provides a precise, neutral studio reference monitoring system for project, commercial and post production studios. The A-20's control amplifier adapts to any production environment by offering control over monitoring depth (from near to far field), wall proximity and even input sensitivity while the speakers magnetic shielding allows seamless integration into today's computer based studios.

- Type Modular, self-powered near/mid/far-field monitor.
- 48Hz - 20kHz frequency response @ 1M
- Peak Acoustic Output 117dB SPL (100ms pink noise at 1M)
- XLR outputs from power amp to speakers
- Matched impedance output cables included.

Amplifier
- Amplifier Power 250W (continuous rms/ch), 400W (100ms peak).
- XLR, TRS input connectors
- Headphone output
- 5-position input sensitivity switch with settings

- -6dB LF Cutoff 40Hz
- 5 position wall proximity control
- 5 position listening proximity control between near, mid and far-field monitoring
- Power, Overload, SPL Output, Line VAC and Output device temperature display.

Speakers
- 2-way acoustic suspension with a 6.5-inch treated paper woofer and a 1-inch aluminum dome tweeter
- Fully magnetically Shielded with an 18-inch recommended working distance

EVENT ELECTRONICS
PS-5 Bi-Amplified Project Studio Monitors

The PS-5s are small format, full-range, non-fatiguing project studio monitors that give you the same precise, accurate sound as the highly acclaimed 20/20 series studio monitors. The use of custom driver components, complimentary crossover and bi-amplified power design provides a wide dynamic range with excellent transient response and low intermodulation distortion.

FEATURES-
- 5-1/4-inch magnetically shielded mineral-filled polypropylene cone with 1-inch diameter high-temperature voice coil and damped rubber surround LF Driver
- Magnetically shielded 25mm diameter ferrofluid-cooled natural silk dome neodymium HF Driver
- 70 watt continuous LF and 30 watt continuous HF amplification per side
- XLR-balanced and 1/4-inch (balanced or unbalanced) inputs
- 52Hz-19kHz frequency response ±3dB
- 2.6kHz, active second order crossover
- Built-in RF interference, output current limiting, over temperature, turn-on transient, subsonic filter, internal fuse protection
- Combination Power On/Clip LED indicator
- 5/8" vinyl-laminated MDF cabinet

KRK V-6 Bi-Amplified Near Field Studio Monitors

These bi-amped studio monitors from KRK supply 90 watts of clean power. Their 6-inch woofer & 1-inch silk dome tweeter ensure consistency from top to bottom with crystal clear highs and a solid bass response.

FEATURES-
- 58Hz - 22kHz frequency response
- 1-inch silk dome tweeter and 6-inch long stroke, polyvinyl woofer
- 30 Watt HF & 60 Watt LF amplification
- Magnetically shielded
- Variable system gain +6dB -30dB
- Neutrik XLR/1/4" TRS combo connector

Also Available- V-8
- 1-inch Silk Dome tweeter and 8-inch Woven Kevlar woofer
- 47Hz - 23kHz frequency response
- 60 Watt high frequency and 120 Watt low frequency amplification
- HF adjust +1dB, Flat, -1dB
- LF adjust -3dB at 45, 50 and 65 Hz

Hafler
TRM-6 Bi-Amplified Near Field Studio Monitors

Offering honest consistent sound from top to bottom, the TRM-6 bi-amplified studio monitors are the ideal reference monitors for any recording environment whether tracking, mixing or mastering. Supported by Hafler's legendary amplifier technology that provides a wide and accurate sound field, in width, height and also depth.

FEATURES-
- 33 Watt HF & 50 Watt LF amplification
- 1-inch soft dome tweeter and 6.5-inch polypropylene woofer
- 55Hz - 21kHz Response
- Magnetically Shielded
- Electronically and Acoustically Matched

Also Available- TRM-8
- 1-inch soft dome tweeter and 8-inch polypropylene woofer
- 45Hz - 21kHz frequency response ±2dB
- 75 Watt HF, 150 Watt LF amplification

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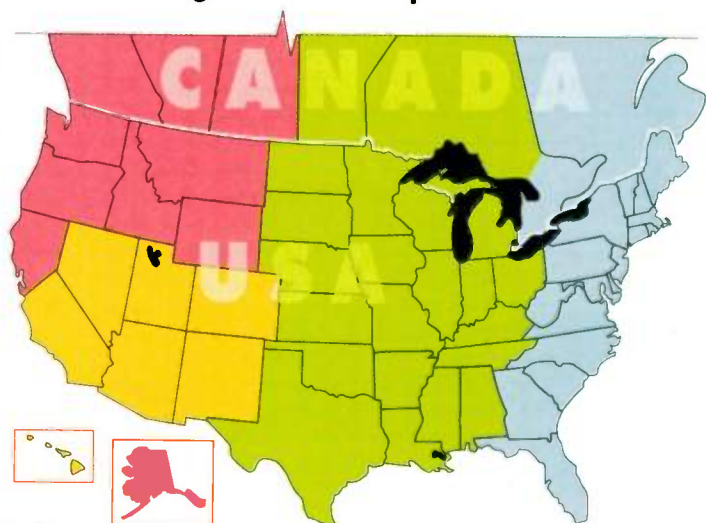
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A GUIDE TO THE COMPANIES AND ORGANIZATIONS MENTIONED IN THIS ISSUE OF ELECTRONIC MUSICIAN

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TC WORKS
ULTIMATE SOFTWARE MACHINES

METRIC HALO LABORATORIES

ANTARES
REALLY COOL STUFF FOR MAKING MUSIC

TC Native Bundle™

TC Electronic hardware effects processors are the cornerstone of many top recording studios. These five plug-ins, featuring the incredible TC native reverb, bring that legendary TC-quality audio processing to your MOTU system desktop. The TC Native Bundle neatly places at your fingertips over 20 years of audio processing R&D, deployed in native 32-bit floating point glory. Incredible sound, well-crafted presets, low CPU overhead, and intuitive controls make these TC Native plug-ins a joy to use.

ChannelStrip™

ChannelStrip is like having a magebuck mixing console inside your PowerMac. Even artists who regularly use top-of-the-line, large format consoles are raving about the "high-end console" sound they get from ChannelStrip. How did Metric Halo do it? By combining 61 standard, fully-automatable audio processing facilities into a single, complete plug-in with 64-bit floating point precision. ChannelStrip is heavily optimized for efficient operation in your MOTU native recording environment, so you can use it throughout your mix. How does ChannelStrip actually sound? Producer Andy Gray-Ling puts it like this: "...I'm absolutely mind-blown. It sounds amazing..."

Antares Microphone Modeler™

Now the microphones you own can sound like the microphones you wished you owned. Mic Modeler allows any reasonably full-range microphone to sound like virtually any other mic. Using patented Spectral Shaping Tool™ (SST) technology, Antares has created precise digital models of a wide variety of microphones, from historical classics to modern exotics, as well as a selection of industry-standard workhorses. Just select which microphone you are actually using and then select what mic you want it to sound like. You can further fine-tune the sound with modeled tube saturation, proximity, windscreen effect, and more. Mic Modeler is an easy, cost effective way to extend your existing mic collection, or to obtain that classic, vintage sound — without the excessive price tag.

MAS COMPATIBLE
MOTU AUDIO SYSTEM

MAS COMPATIBLE
MOTU AUDIO SYSTEM

MAS COMPATIBLE
MOTU AUDIO SYSTEM

Let It Go

All good things come to an end. Fortunately, most bad things do, too. In the world of creative works, the big question is, when is it time to call something finished and put it out into the world? What makes a mix, a sound effect, or a song *done?*

Occasionally you may experience one of life's miracles—you simply feel certain that a work is everything it should be. Those moments are magical, worthy of commemoration and celebration. On the other hand, your schedule or budget may force you to conclude a project. In such circumstances, "finished" often equals "releasable," or, more colloquially, "good enough for rock and roll." That's not a particularly fulfilling way to finish a project, but at times it's your only option. The trick in such cases is to achieve an acceptable level of quality before your time and money run out.

Sometimes the choice is not externally enforced. How many times have you been in an overdub session and heard, "Just give me one more pass at it—I know I can get it better"? I've heard it from others, and I've said it myself.

Mixing is similar, unless automation is available, in which case the situation is worse. You can always tweak to make the vocals a little more even, the bass a little punchier, the panning effect just a smidgen smoother.

Premastering is your last chance to polish or correct your mix. (Remember, the CD mastering house makes a premaster; the pressing plant creates the glass master.) Inherently a finely detailed process, premastering is a wonderful place to get immersed in minutiae. You may actually improve the project with all those adjustments, but you risk losing your perspective, fine-tuning until you can no longer tell whether you're making the music better.

When does the production process finally stop? Well, that's a tough question, because the answer is: only when you say so. There comes a time when you must decide that the project is all it is going to be and then let it go. It's like parents watching their children grow up, but on a much simpler scale.

Part of the problem is that unless you work very quickly, you're likely to have advanced, both artistically and technically, far beyond the material at hand by the time your project is complete. You want your work to reflect your current standards, but it was created to satisfy your former ones. That's a fact you just have to accept.

One of my premastering clients took a test burn home after each session, listened to it intently, and—being a very astute listener and a fast learner—came to the next session with specific adjustments in mind that he wanted made. Though happy with the test burn at each session's end, he always found flaws when he listened to it later. One day, he was struck by the insight that the premaster would simply never sound as good as he thought it should because he had upgraded his sound sources and recording medium since mixing the tracks—and he had learned a tremendous amount in the process. Realizing "that was then and this is now," he sighed and signed off on our final premaster. However, even as he sighed in resignation, he commented that he felt as if a huge load had been lifted off of him.

In contrast, I've seen other clients and friends become stuck on a single project for years because they were unable to do what that client did. Instead they just kept hammering on the same thing endlessly.

Maybe your project is flawed; maybe it's not all that you hoped it would be. Feeling that you can do better motivates you to do so in the future. Look for a point at which you can call your project done, tie it off, put it out, note how your next project can be improved, and have no regrets. ⏺