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April 1999

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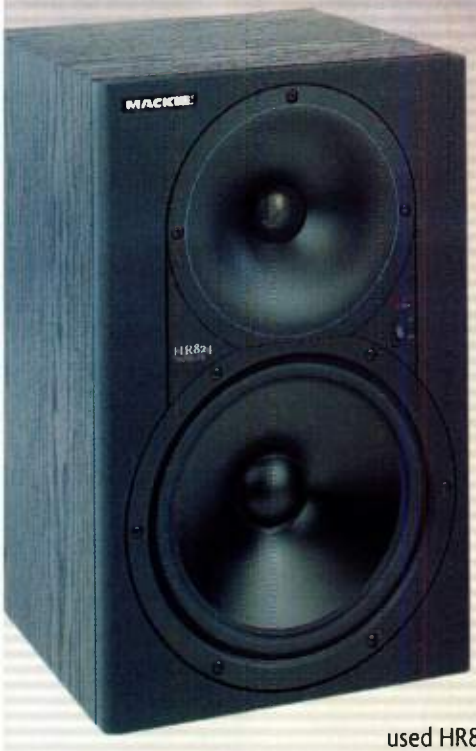


WHY DOES THE RESPONSE OF BECAUSE IT'S

WHERE'S THE EXTRA SUBWOOFER?

Greg Mackie and his team were recently invited to present the Digital 8•Bus to Britain's top engineers and producers in the "A" rooms at two of the world's most famous recording studios. Of course we used HR824 active monitors.

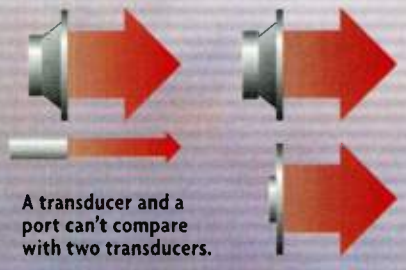
When the presentations were over, many of the veteran engineers were astonished to learn that they had been listening to 8-inch monitors instead of the studio's Big Speakers. Some even so far as to touch the house monitors' 12 and 15-inch cones while the HR824s were playing. They just couldn't believe the bass output from such a compact box.

Last fall we won the pro audio industry's coveted TEC Award for best near field monitor. Modesty prevents us from listing the impressive field of competitors but you'll probably encounter their ads in this magazine.

TIGHT, RESPONSIVE BASS FLAT DOWN TO 39HZ. Reviewers and owner's warranty card responses are unanimous: The HR824 has the most accurate bass they've ever heard from an 8-inch monitor. And the **quality** is as astonishing as the quantity. Fast low frequency transients like kick drum slaps and electric bass notes have a crisp articulation that makes other monitors sound like mush.

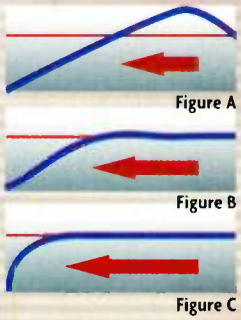
ANOTHER TRANSDUCER INSTEAD OF A PORT.

The more LF transducer cone area a speaker has, the more bass it can produce. But a huge low frequency transducer isn't an option on a compact near field monitor. To augment primary bass output, other monitors resort to using ducted ports that can convert cone movement into extra low frequency air movement. But for optimal output, a ducted port needs to have the **same area** as the low frequency transducer. In other words, an 8-inch near field monitor would need an 8-inch vent. Needless to say, you haven't seen any vents this big on our competitors' near field monitors. When vent size is reduced to maintain compact enclosure size, bass output is compromised. And, forcing a lot of energy out of a couple of small ports can create audible wheezing and whooshing.



A transducer and a port can't compare with two transducers.

Instead, the HR824 adds a large passive transducer with the cone area of another 8-inch woofer. Occupying the entire rear panel of the monitor (see photo below), this ultra-rigid honeycomb laminate piston tightly couples with the 824's active bass transducer. With a combined cone area greater than a single 12-inch woofer, you get exceptionally extended bass without port noise complaint.



Pushing out the curve: redistributing LF energy with synthesized mass.

SYNTHESIZED MASS AND OTHER STORIES.

The cool thing about an active speaker system is that you can basically rewrite laws of physics that otherwise limit passive speaker designers.

A low frequency transducer's free air response graph looks like a bell curve—it's most efficient in the mid band (Fig. A above). To flatten the curve (and extend low bass), you have to proportionally reduce higher frequency output. Acoustic designers use all sorts of tricks to do this—and usually end up with response something like Fig. B.

Rear view: The HR824's electronics conceal an ultra-rigid, honeycomb composite passive transducer.

The most effective way to "shape" an LF transducer's output would be to increase its mass (cone weight). But for designers of traditional passive speakers, adding mass hasn't been a practical option since it would dramatically slow down the woofer's transient response.

HR824 HAVE THE MOST ACCURATE BASS ANY 8-INCH ACTIVE STUDIO MONITOR? REALLY A 12-INCH MONITOR IN DISGUISE.

Because the HR824 is internally powered (active), we could precisely control parameters that normally occur outside of the loudspeaker. Greg and the engineering team were able to create an electronic "symbiotic relationship" between the low frequency transducer's voice coil and its FR Series amplifier voltage output. At mid-band frequencies, the woofer "sees" extra synthetic "electronic mass." This effectively pushes out its lower bass response without compromising its lightning-fast transient response (Fig. C).

MASSIVE POWER THAT WOULD PROBABLY POP A PASSIVE MONITOR.

Punching out crisp bass requires a lotta watts. The FR Series™ high-current bass amplifier module inside the HR824 delivers a solid 150 watts of power with peak output in excess of 250 watts (plus another 100 watts for mid and treble). That's significantly more than any other 8-inch active monitor. Moreover, the HR824's servo coupling and ultra-short signal path put that power to work far more effectively than a passive monitor and a 250-watt stereo amp could.

PART OF A TIGHTLY-INTEGRATED SYSTEM. Our servo bass system is only one contributing factor to the HR824's amazing accuracy.

Internal power amplifiers are "fed" by phase-accurate, low distortion electronic circuitry instead of a crude coil-and-capacitor passive crossover. The HR824's proprietary logarithmic wave guide not only widens treble dispersion but also smooths the midrange transition between high and low-frequency transducers. At the critical 3500Hz crossover point, the alloy HF transducer's output is acoustically the same diameter as the LF transducer's output, thanks to the wave guide's flaring design (refer to the actual HR824 photo on the other page, not our ad folks' fanciful rendering at left).

Indirectly, the HR824's LF transducer even contributes to high midrange accuracy. In many monitors, woofer cone harmonic vibrations bounce around inside the enclosure and then exit through the thin woofer cone. The result: smeared imaging and muddled details. Instead of a chintzy chunk of fluff, the HR824's enclosure is utterly packed with high-density absorbent foam. Cone vibrations go in, but they don't come back out.

DON'T SKIMP. It's amazing how many studio owners will mortgage the farm for money-is-no-object, esoteric microphones... and then monitor on cheap, passive loudspeakers. If you aren't using ACTIVE near field monitors, you're seriously compromising your creative product.

We urge you to visit your nearest Mackie Designs Dealer and seriously audition **all** of their active monitors with some demanding, bass-rich program material. Judge our claims (and those of our competitors) for yourself. We think you'll agree that the HR824 is truly the best of the best.

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I N S I

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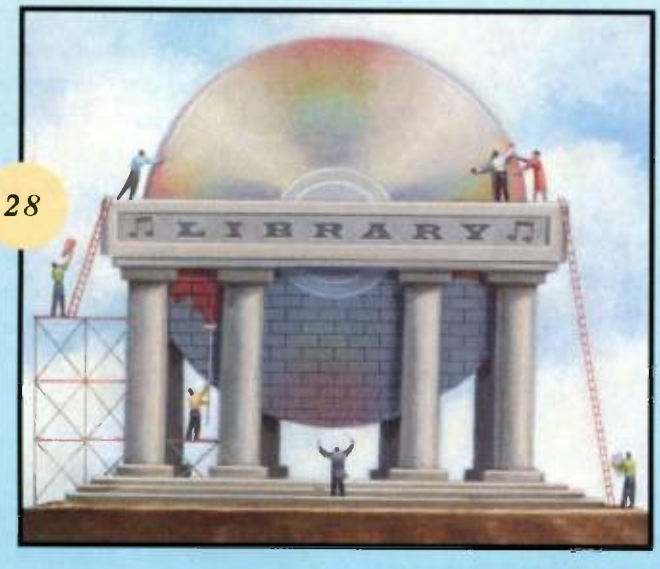

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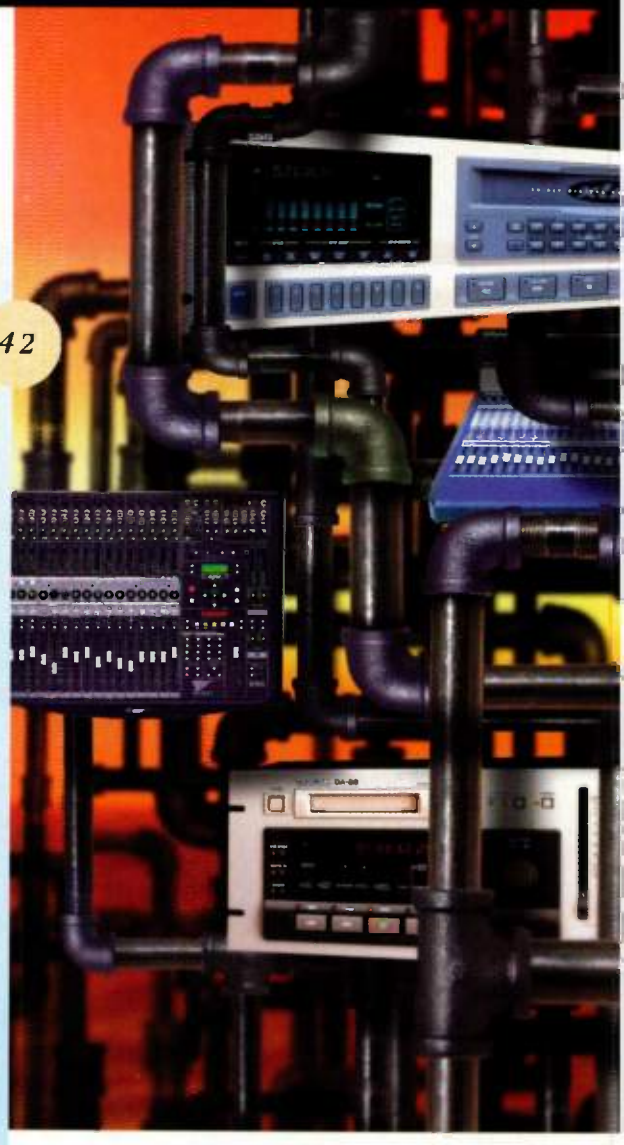

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DEPARTMENTS

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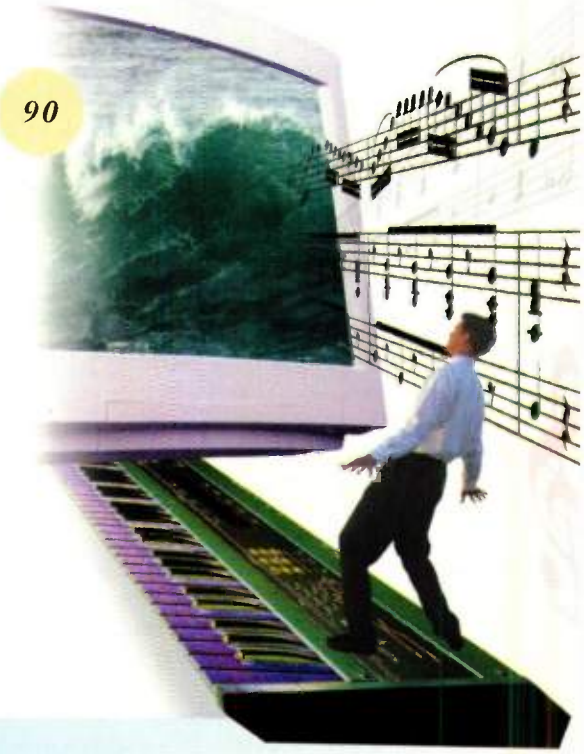
COLUMNS



REVIEWS

Cover: Illustration by Terry Miura

More than Meets the Eye

ANTHONY PIDGEON

This year's Winter National Association of Music Merchants (NAMM) show left me with mixed feelings. Although show attendance appeared healthy enough and *Electronic Musician* was extremely well received, I returned from the show pondering the lyrics of that classic song, "Is That All There Is?"

One reason for my mixed reactions was a lack of compelling new products. I'm not talking about the absence of "breakthrough" products; obviously, the music-technology industry can't be reinvented annually. I'm not even saying that there weren't any cool products; I saw lots of them. But most of them were improved or less expensive versions of existing products. Only a few were truly new.

What's the deal? I think we are witnessing the natural maturation of a high-tech–driven market. To begin with, many new products are software based, which allows developers to quickly release new, improved versions. Software can be tricky stuff, though, so it takes quite a while to develop new high-tech products. As a result, we see upgraded products more quickly than in the past, but all-new stuff takes longer to bake completely.

There's another side to this. We musicians know that a variety of improved—and often less expensive—versions of current products are coming soon. Therefore, we are tempted to wait for the next generation. Sometimes manufacturers invest a lot of labor and money over a long development period, only to discover that customers are playing a waiting game. In such a climate, companies increasingly have to weigh the risks and carefully calculate the right time to move in new directions.

The changing development cycle is not the only reason that truly new goodies were scarce at NAMM. Once upon a time, product releases were timed to coincide with the two NAMM shows and the AES show. Winter NAMM was the primary show for introducing musical instruments and related products. But due to the increasingly international nature of the market and the growing dominance of superstores and large mail-order firms, NAMM is no longer the only major U.S. venue where MI manufacturers show their new products.

Foreign trade shows are part of the equation; if a product isn't ready for Winter NAMM, it can be shown at the Frankfurt Musik Messe. Furthermore, superstores and mail-order behemoths increasingly preview new products at special private showings instead of waiting for trade shows. And the Web makes information and demo software immediately available on a global scale.

What does all this mean for those who attend NAMM expecting to see the shapes of things? The answer is unclear, but the good news might be that much more is happening than we saw at this show. If so, perhaps NAMM is not the bellweather that it once was.

• •

I'm very pleased to announce that "What's New" co-authors and copy-editing colleagues Carolyn Engelmann and Rick Weldon have been promoted to the position of assistant editor. Congratulations to both of them!

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songwriter, musician, producer

ZR-76

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LETTERS

KNIGHT IN SHINING ARMOR

I just finished reading the outstanding feature on Sir George Martin ("To Sir with Love," February 1999). Larry the O does a very respectful and insightful job of getting at the essence of a great man in the far too little space dictated by a magazine format.

I have been a professional in the industry for more than 20 years, as a musician and now as a producer and studio owner. I grew up with the music that Mr. Martin helped realize, and it goes without saying that he has more than influenced all of us in this, shall we say, "interesting" industry. For me, the work of George Martin has been a catalytic inspired surprise; it is truly even more inspiring to discover that such a great artist is also a magnificent human being.

Though Mr. Martin may kiddingly refer to his gentlemanliness as "the kiss of death," it would serve all artists well to respect those around them in a similar fashion, to hold their tongues now and then and possibly learn something. It's rather like the old saw, "It is better to keep your mouth shut and appear stupid than to open it and remove all doubt."

Kudos to Sir George. His influence goes beyond his work. May his retirement be a long and deliriously happy affair.

Jeff Lindeman
jeff@wavemedia.net

ACCESS TIME

The *Desktop Music Production Guide* left me hanging in the article "Choosing a Disk Drive for Audio Recording." Michael Dorian did a fine job explaining access time, but what he neglected to address is: what access time do I need in a hard drive in order to record digital audio glitchlessly? Does regular defragging affect the need for quick access time?

Joe
joemcdonnell@hotmail.com

Author Michael Dorian responds: *Depending on the track count and amount of insert edits (punch ins/outs), crossfades, and edits, access times in the low teens (in milliseconds) will provide the best performance. Regular file optimization can improve the reads by making the data more contiguous. Also, partitioning your drives will reduce seek times (a component of access time) by limiting the distance the heads have to travel to read the data.*

24-BIT/96 KHZ NOT SO NEW

In your 1999 Editors' Choice roundup (January 1999), you write that "*Samplitude 2496*...was the first major multitrack editor to support 24-bit audio at sampling rates up to 96 kHz." In fact, Syntrillium's *Cool Edit Pro* version 1.0 was the first major multitrack editor to support 24/96 when it was released in August 1997, almost a year before *Samplitude 2496* came out. The current version, *Cool Edit Pro* 1.1, is still the only major multitracker to support 192 kHz—it even goes all the way up to 10 MHz. We would greatly appreciate it if you would let your readers know about this.

Bob Ellison, President
Syntrillium Software
bellison@syntrillium.com

TIED TO THE OLD TEXTURE

Please help me. You see, I started sequencing on a Commodore Amiga about ten years ago. I was using the *Texture* program, and I really liked it. I could relate to that format. It was kind of like a drum machine, in that I could sequence sections of a tune and then piece them together any way I wanted to. I've been out of the business, but this year I got involved again. I purchased an 80486 and a Cakewalk program, but I don't relate to that type of sequencing. My question is, can I get *Texture* for the PC, or can you recommend something comparable?

Gil Valentin
gev@swbell.net

Gil—The MIDI sequencer Texture, *from Magnetic Music, was a very popular, pattern-based program that seems to have died many years ago. Today, you won't find many sequencers that bill themselves as "pattern based," mostly because they do both pattern and linear recording. Coincidentally, there are now several "pattern-based" audio programs that allow you to reuse small chunks of audio repeatedly. For example, Sonic Foundry's* Acid *program is a "loop assembly" application that is, in effect, a pattern-based audio sequencer.*

Back to MIDI. I am not aware of any program that is precisely in the mold of Texture, *but you can certainly accomplish many of the same tasks with most modern sequencers. Steinberg* Cubase, *in particular, may be a good choice because of its Editor functions. You'll have no trouble quickly building multilayered, repeating MIDI drum loops with that program. You might also look at MediaTech Innovations'* The Muzical Wizard *and Jazzware's* Jazz++, *two*

The Birth Of A Legend

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the new legend

LETTERS

shareware sequencers that have some unique approaches to generating and using patterns. These programs can be found at www.harmony-central.com, or wherever fine shareware is sold.

On a personal note, I remember Texture *well, as it was my first sequencer. I switched to Voyetra's* Sequencer Plus *after a short time (it, too, is no longer in production), but I used* Texture *for many of my earliest MIDI compositions.—Dennis M.*

MTS TO MIDI

I need your help. I was using Passport *MasterTracks Pro* years ago on my little 80386 PC. I have about 50 files in .mts format, and I no longer have access to *MasterTracks*. Is there a utility or a demo that can convert these old files to a .mid format so they can be imported into Cakewalk *Pro Audio*?

Steve Whitten
steve@nj.innofund.com

Steve—The best way to convert the files is probably to buy MasterTracks Pro *(Mac/Win; $49.99) or* MasterTracks Pro Audio *(Win; $51.99). The Passport software line is still available from new owner G-Vox Interactive (tel. 215/922-0880; fax 215/922-7230; e-mail passport@gvox.com; Web www.gvox.com). By the way, G-Vox is offering some special deals on Passport software to* Encore *owners; check the company's Web site for details.*

Passport products are also distributed online by 4CDs.com (tel. 800/322-8866; Web www.4cds.com/prd.i/pgen/fourcds/0L/music/gvox.html). In addition to MasterTracks Pro *and* MasterTracks Pro Audio, *4CDs.com offers Passport's* Encore *4.2.1 (Win; $399.99),* Encore *4.1.2 (Mac; $399.99),* MusicTime *(Mac; $52.99), and* MusicTime Deluxe *(Win; $52.99).—Steve O.*

ERROR LOG

February 1999, "To Sir with Love," p. 62: Robert Russell Bennett orchestrated the version of "Rhapsody in Blue" featured on George Martin's *The Glory of Gershwin* album.

WE WELCOME YOUR FEEDBACK.

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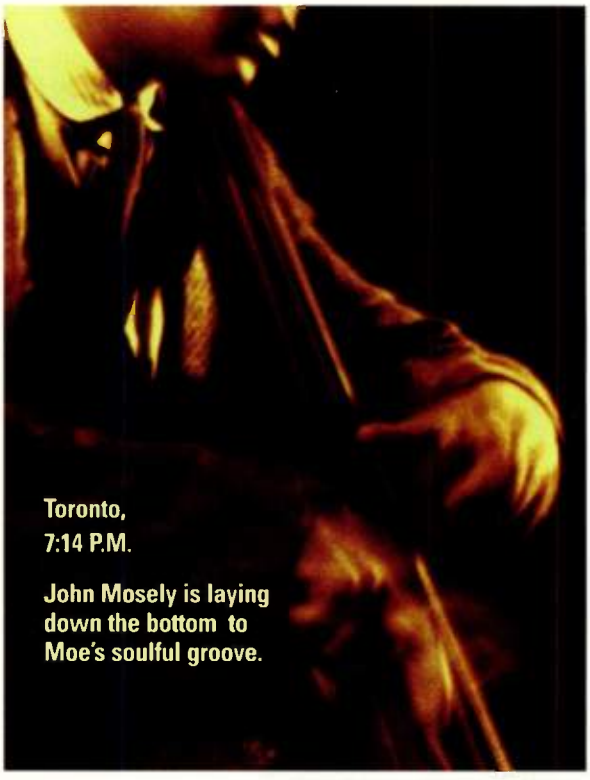

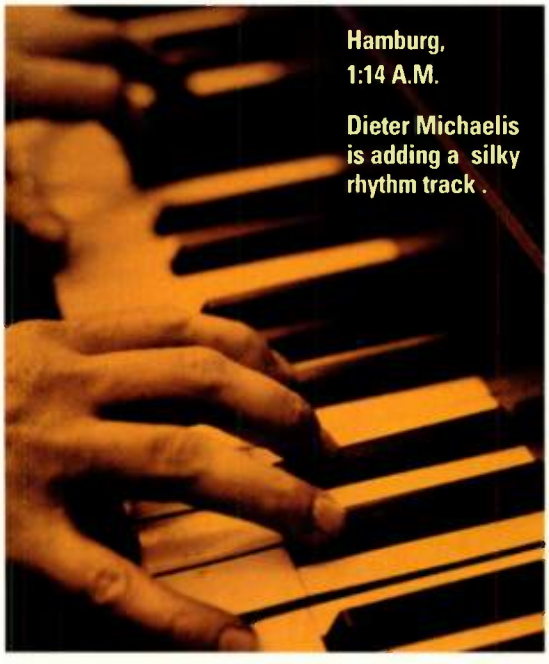

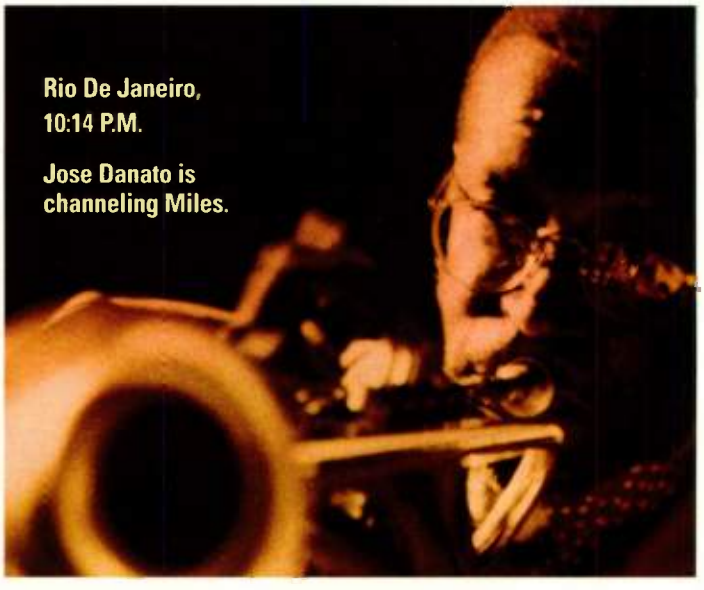

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WHAT'S NEW

By the EM Staff

CURTIS AL-1 MICROPHONE SYSTEM

The AL-1 microphone system ($2,995) from Curtis Technology consists of two vacuum tube mics and a single-rackspace power supply. The mics, each of which has a 2.2 cm diaphragm, are designed to exhibit a polar pattern that is midway between omni- and unidirectional. According to Curtis, the AL-1s can be used successfully on just about any instrument.

Each mic features a 12AU7 tube that has been individually tested and FFT analyzed for output response. According to the manufacturer, with the AL-1 circuit, 12AU7s present less coloration than 12AX7s and respond faster.

The power supply has a ground-lift switch and hosts the XLR inputs. Curtis rates the AL-1's frequency response at 20 Hz to 30 kHz and maximum SPL at 125 dB. The system comes with two shock-mount holders and a one-year warranty that covers parts and labor. Curtis Technology; tel. (815) 399-8453; fax (815) 399-2559.

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STEINBERG QUADRAFUZZ

Steinberg's line of DSP plug-ins now includes the *Quadrafuzz* multiband distortion (Mac/Win; $69). Based on a hardware design by **EM** founding editor Craig Anderton and developed as software by Spectral Design, *Quadrafuzz* splits a signal into four frequency bands and allows you to apply distortion to each band independently. Using the Solo function, you can hear each band individually as you adjust parameters.

Applications abound for this multiband processor. Besides using the software for classic, analog-type, broadband distortion, for example, you could use it only on higher frequencies to produce an exciter-like effect, or apply it to the low end to add warmth and dimension.

You can choose different types of distortion with the Shaping function, and *Quadrafuzz* includes a filter bank for EQ adjustment. Sixteen editable presets, created by Anderton, get you started, including settings for lead guitar, rhythm guitar, and drums.

Quadrafuzz is available in VST, WaveLab, and DirectX formats for PC and in VST format for Mac. The PC version runs on a Pentium 133 or faster with Windows 95/98 or NT and at least 32 MB RAM. The Macintosh version runs on a 604/132 or faster Power Mac with Mac OS 7.5 or higher and 24 MB RAM. Steinberg North America; tel. (818) 993-4161; fax (818) 701-7452; e-mail info@ steinberg-na.com; Web www.us.steinberg.net

Circle #402 on Reader Service Card

GENERALMUSIC SK760 AND SK880

Generalmusic has introduced two new 64-note polyphonic keyboard workstations. The SK760 ($2,495) has 76 semiweighted keys, and the SK880 ($2,995) has an 88-key weighted action. Each synth is 32-part multitimbral and offers 32 layers, 32 splits, and 32 dynamic switches. Each comes with 16 MB of presets, comprising more than 1,000 editable sampled sounds.

Sounds can be edited with 2-pole and 4-pole resonant filters, and you can combine six waveforms in any patch. Twenty-segment envelopes let you program time-variant volume and pan values and filter-cutoff curves.

There are 192 preprogrammed Styles, each consisting of eight tracks; more than 2,000 preset Style Performances provide eight instrumental variations for each Style. Users can create up to 32 custom Styles in RAM.

One of the new Performances in the SK760 and SK880 synthesizers is a classic drawbar organ. In this mode, the instrument's eight sliders function as drawbars, and the speed of the simulated rotary speaker can be controlled on the front panel or with a footswitch.

The SK's 32-track sequencer has a capacity of 16 songs. With four onboard effects processors, you can apply up to two effects per track. The 3D Enhancer algorithm opens up the perceived stereo field for bigger, brighter sound. The Real Audio Synchro Style function lets you synchronize audio sample loops so they will follow tempo and chord changes as you play. A 3.5-inch floppy-disk drive is provided for loading and saving Standard MIDI Files, as well as sample files in Akai, Kurzweil, Sound Designer, WAV, and AIFF formats.

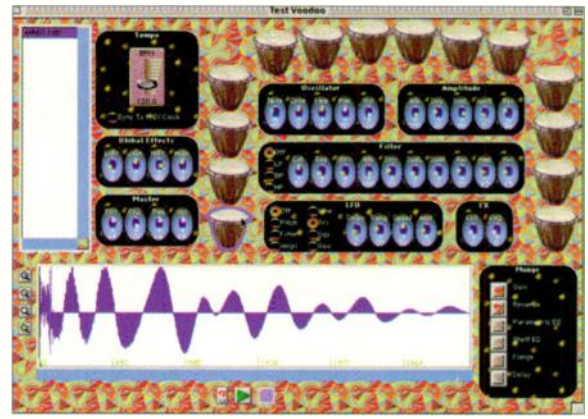
The back panel has ¼-inch jacks for damper and volume pedals and three programmable pedal controllers. You get an 8-pin computer connection and two sets of MIDI In/Out/Thru ports. Generalmusic; tel. (800) 323-0280 or (630) 766-8230; fax (630) 766-8281; e-mail gmail@generalmusicus.com; Web www.generalmusic.com.

Circle #403 on Reader Service Card

BITHEADZ VOODOO & OSMOSIS

BitHeadz introduces *Voodoo* (Mac/Win; $199), a software drum machine that plays samples and Standard MIDI Files. The program can import and export samples in AIFF, SDII, WAV, and BitHeadz's Unity DS-1 formats. Each *Voodoo* file accommodates an entire virtual drum kit, with MIDI control data and sequenced patterns, all in one window, and you can have several files open at once. Programmed pads and patterns can be triggered from the computer keyboard, a MIDI controller, or a separate MIDI application.

When you drag a MIDI file onto *Voodoo*'s playlist, you can display its tempo, volume, effect 1 and 2 sends, and MIDI sync parameters. To create your own drum kit, you drag samples or SMFs onto the 14 onscreen drum pads. When you click on a pad, the window displays its editable parameters for pitch, volume, filter, pan, Velocity, envelope, two effects processors, and more. Each pad can layer up to four samples, which can be switched via Velocity or Control Change messages.



Clicking on a pad also brings up a graphic waveform display where you can assign loop points; cut, copy, and paste portions of the sound; and apply DSP functions. Available effects include shelving or parametric EQ, gain, reverse, flange, and delay. Once edited, sound files can be resaved to disk.

Voodoo will run on a Power Mac 601/120 MHz or faster with Mac OS 7.6.1 or higher and at least 32 MB RAM, or on a Pentium 200 or faster running Windows 95/98, at least 32 MB RAM, and a DirectX-compatible sound card.

Also new from BitHeadz is the *Osmosis* sample-conversion utility (Mac; $199). With *Osmosis,* you can convert any Akai S1000/S3000-format sample to AIFF, Sound Designer II, WAV, or Unity DS-1 format. It will also convert multisamples and programs (including filter settings and envelopes) between Akai S1000/S3000, Digidesign SampleCell, and Unity DS-1 formats. The software allows you to audition files before converting them. BitHeadz; tel. (831) 465-9898; fax (831) 465-9899; e-mail info@bitheadz.com; Web www.bitheadz.com.

Circle #404 on Reader Service Card

FOSTEX VM04

Everyone would like to see more affordable digital gear, and to that end, Fostex offers the new VM04 compact digital mixer ($299). Not lacking in features, the 12 x 8 x 2-inch unit operates at 44.1 kHz and has 20-bit A/D converters with 24-bit internal processing.

The VM04 has four stereo input channels with both ¼-inch mic/line-level and ¼-inch line-level inputs. The stereo outputs are on ¼-inch connectors. S/PDIF coax digital output is also included. Each channel has a 30 mm fader, 2-band shelving EQ, pan, and an effects send. A jog wheel adjusts the panning position and EQ values.

The 20 onboard digital effects include reverb, delay, chorus, and flange. These can be assigned prefader or postfader and have real-time parameter control via the jog wheel.

The VM04's scene automation is valuable for live applications. The mixer can store 20 scenes, including input and effects-send levels. Scenes are numbered and can be named by the user. With an optional footswitch controller, you can quickly scroll through all the scenes in memory.

A backlit LCD on the front panel displays input and output meters, as well as effects parameters and scene names and numbers. Also on the front is a ¼-inch headphone jack with its own level knob. Fostex Corporation of America; tel. (562) 921-1112; fax (562) 802-1964; e-mail info@fostex.com; Web www.fostex.com.

Circle #405 on Reader Service Card

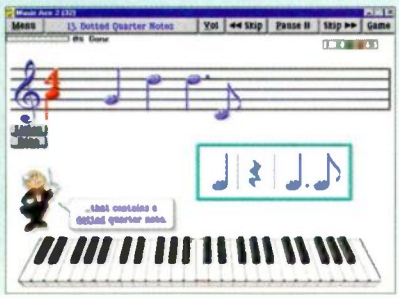
CAD CM 17

Looking for an all-purpose condenser mic that won't break your budget? Targeted at both sound reinforcement and recording, the sleek, all-black CAD CM 17 ($179) electret condenser microphone is designed to provide high gain while producing very little self-noise. CAD claims that the CM 17's transparent sound and ability to handle transient signals make it a good overhead mic for drums, as well as a good choice for recording vocals and for miking acoustic guitar and other acoustic instruments.

The cardioid microphone can be powered by a 1.5V AA battery or by 9 to 52V phantom power. Maximum signal handling is rated at 130 dB SPL, frequency response at 30 Hz to 18 kHz, and signal-to-noise ratio at >50 dB. A microphone clip is included. CAD Professional Microphones; tel. (888) 702-7075; fax (440) 593-5395; e-mail sales@ctiaudio.com; Web www.cadmics.com.

Circle #406 on Reader Service Card

GET SMART



HARMONIC VISION

Beginning musicians, whatever their age, will benefit from developing an understanding of fundamental music theory. *Music Ace 2* (Mac/Win; $49.95), the second in a series of educational titles from Harmonic Vision, is designed to be a fun and easy way to learn.

With *Music Ace 2,* students work at their own pace through 24 lessons that cover topics from note values and rhythm to key signatures, intervals, and scales. Each lesson ends with a game, providing a fun way to reinforce new concepts. The program tracks students' progress through the lessons and records game scores.

Students have the opportunity to apply their new knowledge creatively with the Music Doodle Pad. Here they can write and save original compositions using the mouse or a MIDI keyboard. The Jukebox feature plays and displays well-known songs, and the student can play along or modify the songs onscreen.

For music educators, *Music Ace 2* is available in multiple-user and network versions; contact Harmonic Vision for prices. You can order a demo CD-ROM for $4.95. The Music Doodle Pad is available as a stand-alone program with a teaching guide for $24.95.

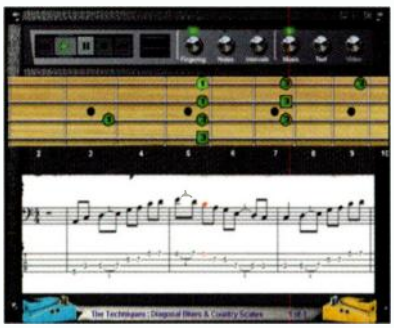
Minimum PC requirements for *Music Ace 2* are an 80486/33 MHz with 8 MB RAM for Windows 3.1 or 16 MB RAM for Windows 95/98. For Macintosh, the program requires a 68040/35 MHz CPU or faster with 12 MB RAM for Mac OS 7.5 or 24 MB RAM for Mac OS 8.0. Harmonic Vision; tel. (800) 644-4994 or (847) 467-2395; fax (847) 467-3008; Web www.harmonicvision.com.

Circle #407 on Reader Service Card

PLAYPRO

With PlayPro's *Interactive Bass* software tutorial package, you can plug your bass straight into your PC's sound card and commence learning, practicing, jamming, or recording. The Standard version (Win; $69) provides two CD-ROMs and a ¼-inch-to-⅛-inch adapter, and the Deluxe package (Win; $99) includes the CD-ROMs, the adapter, two audio CDs, and a 164-page illustrated book that covers the same territory as the CD-ROM.

The CDs, which contain 150 exercises, examples, and lessons, have material for both beginners and more advanced players. Topics include instruction on playing scales, articulating



notes, tuning, slapping, muting, positioning your hands, maintaining your instrument, and much more.

Interactive Bass's Comp-U-Pare function registers the signal coming from the bass into the sound card. With Comp-U-Pare, you can play into your computer and compare your playing with the CD-quality example files, without using a MIDI pickup or other interface. The program grades the results and offers advice based on your playing. If you wish to change the tempo of an example or lesson, you can choose a MIDI version instead of audio. This also lets you transpose, loop bars, and change relative volumes of instruments. The included virtual tuner lets you tune your bass from within the program. A section titled "50 Famous Feels" contains digital audio tracks of bass parts covering a wide variety of musical styles.

Interactive Bass requires at least a Pentium PC running Windows 95, 98, or NT 4.0 and 16 MB RAM. PlayPro Software, Inc.; tel. (408) 969-0800; fax (408) 969-0200; e-mail playpro@.com; Web www.playprosoft.com.

Circle #408 on Reader Service Card

REED KOTLER

Reed Kotler Systems' *Transkriber* (Mac/Win; $49.95) can accept digital audio from your computer's CD player or sound card and adjust its playback speed in real time while maintaining the original pitch. The program is designed to help users transcribe prerecorded music or learn solos and other parts from favorite songs.

With *Transkriber,* you can adjust the playback speed to virtually any fraction of the original, such as ½, ⅔, ¾, ⅛, and so on, down to 1/28 of the original speed. The program does not need to create an intermediate sound file. You can also adjust the incoming audio's speed and pitch while listening in real time, and a transpose function lets you adjust the pitch of the audio up or down by as much as an octave.

The program's other features include a looping function and phrase selection, both of which can be enacted during playback. You get a vocal-elimination function and filter presets designed to bring out particular instruments (including alto sax, bass, guitar, and trumpet). A variable pitch generator provides reference notes.

Transkriber requires a Pentium processor or a Power Mac with 20 MB RAM for real-time performance. Reed Kotler Systems; tel. (408) 730-9557; fax (408) 733-9547; e-mail reed@reedkotler.com; Web www.reedkotler.com.

Circle #409 on Reader Service Card

KORG

N5EX

MUSIC SYNTHESIZER

61 KEY SYNTHESIZER SYSTEM
64 VOICE POLYPHONY
ARPEGGIATOR
DYNAMIC STEREO EFFECTS

DOES MORE. SOUNDS BETTER. COSTS LESS.

OVER 1700 UNBELIEVABLE KORG SOUNDS

The affordable N5EX packs 18 Mbytes of sound-generating power, delivering 1,671 sounds and 39 drum kits—sounds made famous in legendary Korg keyboards like the M1, 01/W, Trinity and SGproX. Plus the stereo piano, electric piano, organ and clav found in the N5EX's siblings, the N1 and N1R. Not to mention a pair of digital stereo multi-effect systems—each with 48 excellent effects—including resonance filter, chorus and delay.

DOUG BECK

INCREDIBLE REALTIME CONTROL

Whether you're gigging, recording, or just having fun, the N5EX's four front panel knobs provide real-time control of up to 16 sound parameters, including Attack/Release Time, Filter Cutoff and Effect Modulation. And with 32 memory locations in which to save favorite Performance settings, you can call back split, layer, arpeggiator and knob settings with the touch of a button.

FEATURES GALORE...

The N5EX is loaded with everything you'd expect—like 64-note polyphony and 32-part multi-timbrality. And some very cool stuff you wouldn't—like polyphonic portamento and a MIDI-syncable arpeggiator.

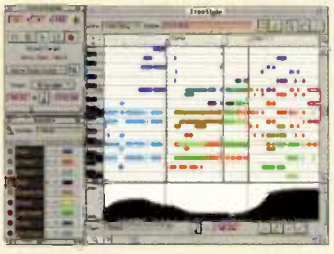
MORE MODELS TO CHOOSE FROM.

All the fabulous features of the N5EX come standard in the N1, a weighted-action, 88-key marvel that marries real piano feel and response to NTH level performance. And the N1R squeezes all that power and utility into a single rack space module.

DISCLAIMER: SHARP DUDS AND A GROOVY ATTITUDE HAVE NOT BEEN PROVEN TO ENHANCE KEYBOARD PERFORMANCE (BUT YOU GOTTA ADMIT THEY LOOK PRETTY BITCHIN')

MIDI-PHILES LOVE THE N5EX

The new N5EX fully supports General MIDI and includes GS & XG sound maps. That means it can speak the language of all commercially available



GM, XG, OR GS- THE N5EX PLAYS 'EM ALL BACK. PLUS, BOTH THE N5EX AND N1 COME BUNDLED WITH MARK OF THE UNICORN'S *PRESTYLE* LE SEQUENCING AND UNISYN EDITOR/LIBRARIAN SOFTWARE—A $$$ VALUE!

MIDI file information. It also comes with a built-in computer interface that's both PC and MAC compatible.

SURFERS WELCOME...CHECK OUT WWW.KORG.COM

For more information on the N5EX and the rest of Korg's outstanding line of music products, make a point to drop in at Korg's home on the cyber-range.

*Produces/Remixer Doug Beck's remix credits include Salt-n-Pepa, The Rolling Stones and Shania Twain.

KORG
Super sonic.

Digital Control With the Human Touch

The Spirit 328 represents a **new way of thinking** in digital console design, bringing all the functionality and sonic excellence of digital mixing to all audiences. With its unique **console-based interface**, the Digital 328 finally bridges the gap between analog and digital mixers, retaining the **spontaneity and ease of use** of an analog console yet providing all the advantages of digital, such as instant Total Recall of all digital parameters, Moving Fader Automation and Onboard Lexicon Effects.

Simply put, the Spirit **Digital 328** is the most advanced analog 8-bus you have ever seen combined with the easiest digital console you have ever used.

328's E-Strip may be configured as...

...a horizontal input channel with full EQ, Aux and Pan facilities

...or a row of 16 Auxes or FX Sends for the 16 faders below the E Strip...

...or a set of rotary level controls for the Tape Returns.

All the Digital I/Os you need as standard

Most digital mixers don't include digital multitrack I/Os, which means that to get digital recording and mixdown you have to buy extra, expensive I/O options. In contrast, Digital 328 includes **two Tascam TDIF™ and two Alesis ADAT™ optical interfaces as standard**, allowing you to record 16 tracks entirely in the digital domain, straight out of the box. As you would expect, we've also included a pair of **AES/EBU and SP/DIF interfaces** assignable to a wide range of inputs and outputs, including group and auxiliary outs. In addition, a third optical output may be used as a digital FX Send or as eight Digital Group Outs. All in all, there are 28 Digital Outs on 328 plus 20 Digital Returns, providing enough flexibility for the most demanding applications.

2 Onboard Lexicon Effects Units

Only 328 can offer the world's premier name in studio effects onboard - Lexicon. Two separate effects units are included, offering a full range of reverbs, choruses, delays and flanges, all with fully editable program and parameter settings.

42 Input/8 Bus Configuration

For a mixer with such a small footprint, Digital 328 packs an extraordinary number of inputs. Sixteen full spec. analog mono mic/line channels - each with its own balanced XLR connector, dedicated insert point and access to phantom power - come as standard, along with five stereo inputs.

With the 16 digital tape returns on 328's TDIF™ and ADAT™ optical interfaces, there's a maximum of 42 inputs. Every input is fully routable to any of the 8 groups and has access to the full complement of 328's parametric EQ, signal processing, onboard effects and auxiliaries.

As easy to use as your current analog console

Although most digital mixers offer an amazing array of functions, it can often be a nightmare to access them.

In contrast, we've designed Spirit 328 to operate like your old analog 8-bus console, and not like a computer with faders. You can practically take it out of its box and get started without even opening the manual! Unlike other digital mixers, there's instant access to any channel, group or master feature with one button press, and you can see that feature's status from the front panel without having to rely on an LCD display. Access is so immediate that you could even use 328 as a live console.

The key to it all is Spirit 328's unique "E-strip", the lighter-colored bank of encoders and switches that runs across the center of the console. Simply select a channel and the E-strip immediately becomes a "horizontal input channel" with instant access to all that channel's EQ, aux sends, channel pan and routing. Alternatively, press any button in the rotaries section above the E-strip and the encoders change to become a channel pan, auxiliary send or Lexicon effects send for each channel.

Select a fader bank to display mic/line input faders, tape returns faders or group and master faders and that's it; no delving through level after level of LCD menus to find the function you want, no delays in making alterations and no need to study complicated EQ curves. With 328, everything you need is immediately accessible from the front panel of the console - giving you the freedom to let your ears decide.

If you want the functionality of a digital console but the usability of your old analog 8-bus, then Spirit 328 is for you.

Onboard Dynamics

Digital 328 includes two mono or stereo signal processors which can be assigned to any input, output or groups of ins or outs. Each processor provides a choice of compression, limiting and gating, as well as combinations of these effects.

digital

three two eight

Comprehensive EQ

All of 328's mic/line, tape return and stereo inputs have access to three bands of fully parametric EQ, designed by British EQ guru and co-founder of Soundcraft, Graham Blyth. A man with over five million channels of his EQ designs in the field, Graham has brought 25 years of Soundcraft analog EQ circuit experience to bear on Digital 328. If you want the warm, musical sound of real British analog EQ, with proper low, mid and high frequency bands (rather than the low resolution 20Hz - 20kHz bands found on some consoles), look no further.

Unparalleled Sonic Spec

Garbage in, garbage out! It doesn't matter whether the console is digital or analog - if you have poor mic preamps, your sound will be compromised. That's why 328 includes Spirit's acclaimed UltraMic+™ padless preamps, giving your input signals the cleanest, quietest start of any digital mixer on the market. With 66dB of gain range and a massive +28dBU of headroom, they offer an extremely low noise floor and are virtually transparent. Spirit 328 is 24-bit throughout, with 56-bit internal processing; your signal hits the digital domain through state-of-the-art 24-bit ADCs with 128 times oversampling, guaranteeing that it maintains its clarity, while 24-bit DACs on all main outputs equal this sonic integrity should you wish to return your signal to the analog world.

Moving Fader Automation

All of Digital 328's 100mm faders (including the master) are motorized to allow current channel, tape return, group and aux master levels to be viewed at a glance.

All Parameters Instantly Recallable

In addition to level automation, every other digital parameter of 328 is instantly recallable, allowing snapshots of the entire console's status to be taken. Up to 100 of these "scenes" may be stored internally and recalled either manually, against MIDI clock, or against MTC or SMPTE. Alternatively, every console function has been assigned its own MIDI message allowing dynamic automation via sequencer software.

Easy To Edit - Direct From The Control Surface

The majority of 328's input and routing parameters may be edited from the control surface without resorting to the console's LCD. Settings and levels may be copied and pasted from one channel to another with just two button presses and, using 328s query mode, the routing or assignment status of every channel on the console may be viewed instantly simply by selecting the function (such as Group 1 or Phase Reverse) you want to question. In addition, with 328's Undo Redo function located in the master section, editing is entirely non-destructive, allowing you to A/B test new settings with previous ones.

Grows With Your Needs

Two Digital 328s may be digitally cascaded, giving you up to 84 inputs at mixdown and 32-track digital recording capability.

Full Metering & Monitoring Options

All of the mic/line inputs, tape return inputs, group and master levels may be monitored per bank via Digital 328's 16 10-segment bargraph meters. Additionally, 328's onboard dynamics processors may be monitored using the console's master meters. Any input may be solo'd using AFL, PFL or Solo-in-Place.

Timecode & Machine Control

Digital 328 reads and writes MTC and reads all SMPTE frame rates, with a large display instantly indicating current song position. Store and locate points are accessible from the console's front panel, with 328's transport bar controlling a wide range of devices including Tascam and Alesis digital recorders.

Software Upgradeable

328's open architecture means that any functional improvements and software upgrades can be made easily available off Spirit's website. 328 Mixer Maps for popular sequencing software packages are also available free of charge.

Add-On Module Options

To meet the needs of a variety of users, there are three module options:

8 Channel Analog I/O Interface

Connecting to the TDIF ports, 16 phono connectors provide eight analog group or direct outs and eight analog inputs for tape returns 17-32. Two interfaces may be connected allowing 16 track analog recording or access to 16 more sequenced keyboard or sampler inputs.

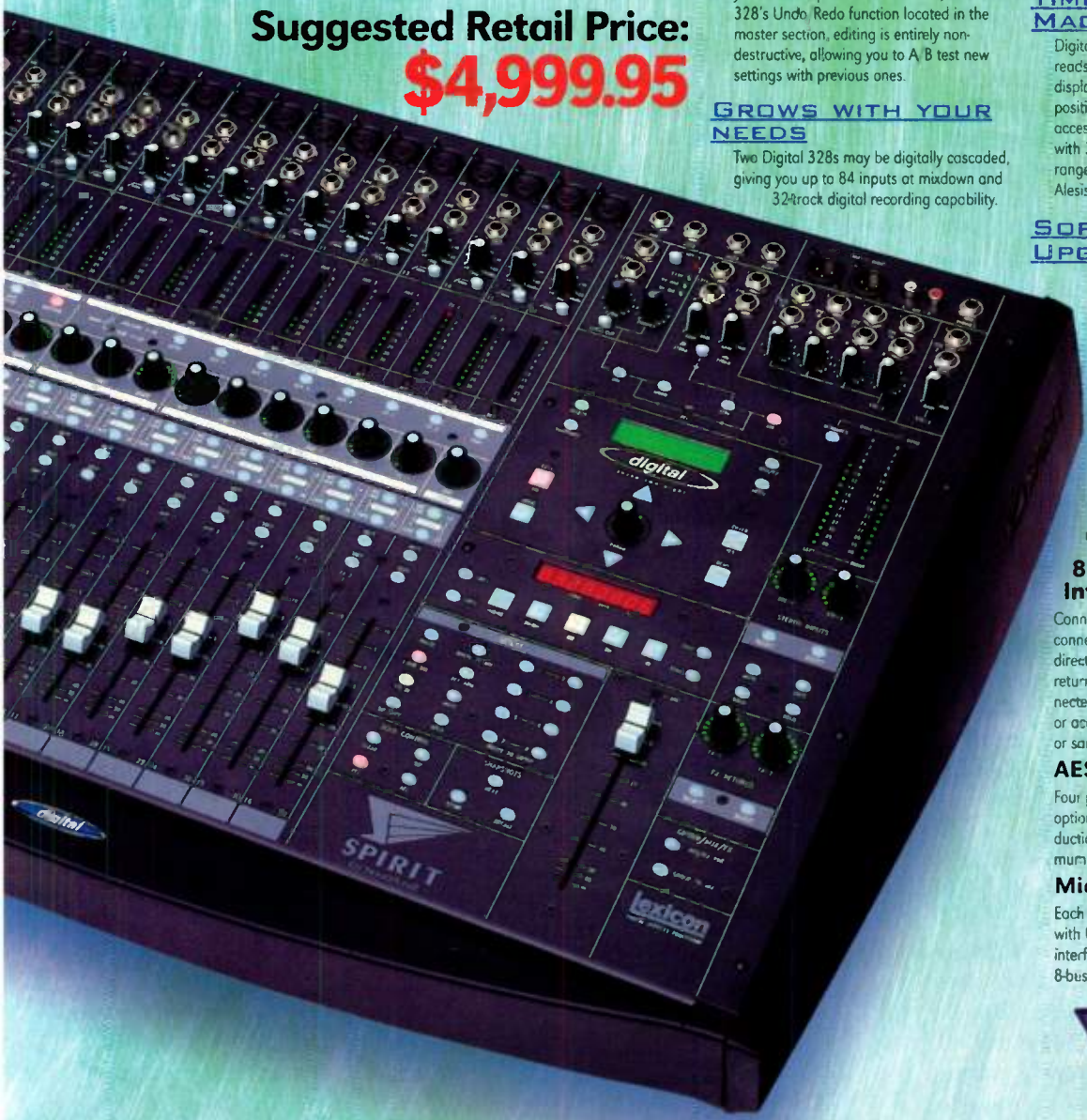
AES/EBU interface

Four pairs of AES/EBU connectors allow optional digital interfacing to hard disk production systems such as Pro Tools. A maximum of two interfaces may be connected.

Mic Pre-Amp Interface

Each interface provides eight XLR mic ins with UltraMic+ preamps. Connecting two interfaces turns 328 into a 32 mic input, 8-bus mixer for PA or theatre applications.

Suggested Retail Price:
$4,999.95



SPIRIT by Soundcraft

A Harman International Company

Spirit By Soundcraft, Inc. • 4130 Citrus Ave., Suite 9 • Rocklin, CA 95677
Toll-free: (888) 459-0410

www.spiritbysoundcraft.com

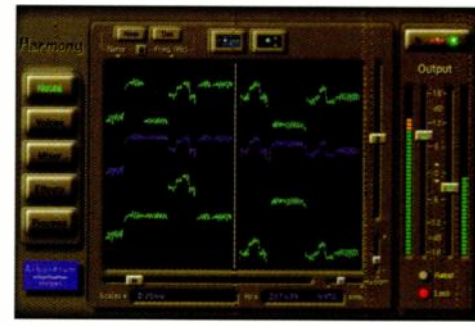
circle #509 on reader service card

KEY CHANGES

Musicians interested in vacuum tubes can find information on where to find vacuum-tube manufacturers, tube-related equipment, and repair services for tube products from the *1999 World Tube Directory* ($7.95), published by **Audio Amateur, Inc.** The company also publishes the bimonthly *Glass Audio* magazine for tube enthusiasts, in addition to such books as *Audio Electronics, Speaker Builder, Voice Coil,* and the *Loudspeaker Industry Sourcebook.* You can order these publications by calling (603) 924-9464 or via e-mail at custserv@audioxpress.com...**Quested** monitors will be distributed in the United States by **Q.usa.** For information about Quested products, call (608) 251-2500...The **International Wind Synthesis Association** has been formed. This nonprofit organization is "dedicated to the advancement of all aspects of wind synthesis, including wind-controller and breath-controller performance, technology, hardware, and software." For more information, go to the IWSA Web site at windsynth.org...**Wave Mechanics** has announced the release of the UltraTools plug-in bundle ($895) for Digidesign Pro Tools TDM systems. The bundle consists of Wave Mechanics' *Pure Pitch, PitchDoctor,* and *SoundBlender* plug-ins...**E-mu** and **BitHeadz** have signed a letter of intent to begin work on a Macintosh-based sampler/synthesizer that will combine the PCI digital audio card from E-mu's Audio Production Studio with software to be based on BitHeadz's *Unity DS-1* software sampler. The hardware/software system is expected to ship later this year with a price of $699...British sample-CD developer **Zero-G** has announced that **East West Communications** will be Zero-G's exclusive U.S. distributor.

—Rick Weldon

ARBORETUM HARMONY



Arboretum's *Harmony* (Mac; $349) is a stand-alone, formant-based pitch-processing and harmonization application designed to let you fix pitches, create new vocal lines, and create new, independent parts with just a few mouse-clicks. Separate formant and pitch controls allow you to create everything from subtle accentuation of the original signal to outlandish vocal treatments. A graphic editor allows you to draw in pitch and harmony changes.

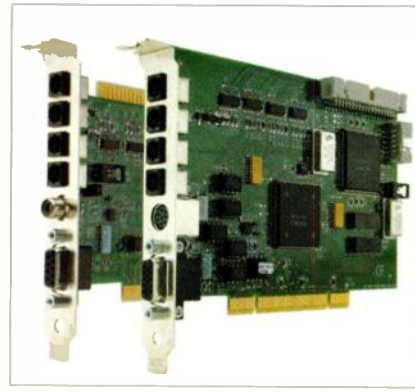
Harmony can generate up to eight separate harmony voices offline, and it will preview, in real time, as many voices as your CPU can process. It also features pitch correction that preserves the vocal formant for natural-sounding pitch shifting. Pitch correction can be applied using a number of scales; scale and harmony libraries are included with the program, and all scales are user adjustable. You can control the sonic characteristics of each voice independently by adjusting the formants, pitch-tracking accuracy, and pitch offset. You can also control volume and pan position independently for each voice.

Harmony requires a 604/200 or faster Power Mac running Mac OS 8.0 or higher for real-time preview of two or more harmony voices; slower Power Macs can process audio files offline. Arboretum offers direct driver support for Digidesign's Audiomedia II and III cards and Korg's 1212 I/O. Arboretum Systems; tel. (800) 700-7390 or (650) 738-4750; fax (650) 738-5699; e-mail info@arboretum.com; Web www.arboretum.com.

Circle #410 on Reader Service Card

FRONTIER DESIGNS DAKOTA

Frontier Designs has released the Dakota ($649), a PCI audio card designed for use with digital tape machines, digital mixers with optical I/O, and Frontier Designs' Tango and Zulu A/D and D/A converters. The card has two pairs of optical digital I/O ports that can be used in either ADAT Optical (for up to 16 channels of ADAT I/O) or S/PDIF format. A breakout cable provides 9-pin ADAT sync and stereo S/PDIF I/O on coax connectors. The Dakota can also import digital audio from a computer's CD-ROM drive via an internal cable.



The Dakota supports 16-, 20-, and 24-bit digital transfers. It generates 44.1 and 48 kHz sample rates and can lock to incoming digital audio at any sample rate between 39 and 51 kHz. The card features Frontier Designs' SoDA (SMPTE on Digital Audio) technology, which lets you use any digital audio input or output for SMPTE time code.

Two pairs of MIDI connectors allow the unit to act as a 32-channel MIDI interface. A copy of Syntrillium's *Cool Edit Pro SE,* drivers for Windows 95 and 98, and an application for controlling the Dakota's features, such as audio routing and SMPTE levels, are included.

Frontier Designs also introduced the Montana ($249), an expansion card for the Dakota that adds two more pairs of optical I/O. An RCA input allows your PC to sync with an external NTSC or PAL video signal or with external word clock. The Montana also has a 9-pin ADAT sync output. Frontier Design Group; tel. (800) 928-3236 or (603) 448-6283; fax (603) 448-6398; e-mail sales@frontierdesign.com; Web www.frontierdesign.com.

Circle #411 on Reader Service Card

Up the value of your mic locker without dropping another nickel into it.

Pit an amateur with $10,000 worth of microphones against the pro with a $1,000 mic and the better recording will emerge from the latter corner, pretty much every time. How? Quite simply, pros use their ears.

The best engineers lean hard and often on their monitors to listen critically and tweak accordingly — placing mics, setting gains, adjusting eq — getting everything dialed in before the first take.

Furthermore, a detailed, accurate monitoring system is what patches flesh and blood into the electrical system and provides a clear window to the sound at every step of the recording process. This is precisely why the best in the business agree that their monitors are the single most critical piece of hardware in the studio.

All of which is supported by the thinking behind NHTPro active studio monitoring systems. From exactly tailored near and mid-field radiation characteristics to totally new and innovative driver control circuitry, NHTPro systems embody all we know about high performance sound reproduction. Which, as anyone who has had the pleasure of hearing them will attest, is considerable.

To find out more, please visit our website @ NHTPro.com or call toll free 1-877-4NHTPro (464-8776).

A-20 Monitors

NHTPro's reference monitoring system. The ultimate in high-end sonics for project, production and post studios. The A-20 combines remarkable precision, detail and dynamics with real functionality and useability.

A-20 Control Amplifier

Configuration:
Modular self-powered near/mid/far-field monitor.
2-way 6". acoustic suspension.

Amplifier:
250w/ch continuous. 400w peak.
<.4% THD 100Hz - 10kHz.
Hum/noise: <10 dB SPL

Monitors:
Response: 40Hz - 20kHz.
Peak output: 117dB.
Magnetic shielding.

NHT PRO

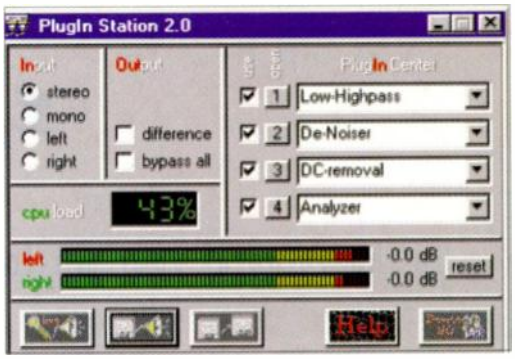
www.nhtpro.com

ALGORITHMIX SOUND LAUNDRY

German manufacturer Algorithmix has created Sound Laundry 2.1 ($249 on CD; $234 if purchased on the Internet), a bundle of real-time digital audio restoration software tools for Windows-based PCs. The software can process with 80-bit internal resolution and is designed for fast, easy, and high-quality restoration of sound files and real-time input.

The bundle consists of five plug-ins and a host program, the *Plug-In Station*. Within the *Plug-In Station*, you determine which plug-ins to use and in what order. You can load WAV files for offline processing, or you can process files or incoming audio in real time and send the results out through your sound card. A Difference feature allows you to hear the noise extracted from your original sound file.

The five plug-ins include *De-Scratcher*, which is best suited to cleaning up sound from vinyl records, although Algorithmix claims it can also remove digital clicks and buzz created by switches and rectifiers. It features onscreen sliders for adjusting declick and decrackle, as well as De-Plop, which reduces unwanted low-frequency impulses. *De-Noiser* offers control over threshold and reduction amount for its noise reduction, and *High-Lowpass Filter* is intended for ridding audio of hiss and rumble. The high- and lowpass filters are designed to avoid creating phase distortion, and their slopes are continuously adjustable up to 24 dB/octave.



The *Analyzer* is designed for before-and-after comparisons when using *De-Noiser* and *Filter*. It shows three curves simultaneously: the frequency-response curves of the signal before and after processing, and the frequency-response curve of the removed noise (when used with *De-Noiser*) or the filter's characteristics (when used with *Filter*). Finally, *DC-Removal Filter* is designed to increase the dynamic range of the incoming signal by correcting the DC offset that can occur when using low-quality A/D converters.

The bundle requires at least a 133 MHz Pentium for real-time operation. Algorithmix GmbH; tel. 49-7741-919300; fax 49-7741-672257; e-mail info@algorithmix.com; Web www.algorithmix.com.

Circle #412 on Reader Service Card

DEMETER EQ-1

Demeter Amplification's dual-channel EQ-1 tube parametric equalizer ($1,295) is a 1U rack-mount unit that combines 12AX7a-based tube circuitry for amplification with a solid-state parametric circuit that controls the gain of the tube at selected frequencies.

The unit has three bands of EQ per channel. The low-frequency band is variable from 20 to 200 Hz, the mid band from 200 Hz to 2 kHz, and the high band from 2 to 20 kHz. The Q is variable from 0.8 to 3.6 octaves, and each band provides up to 15 dB of boost or cut.

The EQ-1 offers a Link switch for 6-band mono operation and has global bypass and a gain-overload indicator. Gain is adjustable up to +10 dBV. The channel inputs and outputs are on balanced ¼-inch TRS and XLR jacks, and there is a ¼-inch mono output on the front panel.

Frequency response is rated at 20 Hz to 20 kHz, THD at 0.035% (at 1 kHz, +4 dBu), and signal-to-noise ratio at -91 dB. The operating voltage is switchable between 230 and 115 VAC. Demeter Amplification; tel. (818) 994-7658; fax (818) 994-0647; e-mail info@demeteramps.com; Web www.demeteramps.com.

Circle #413 on Reader Service Card

FURMAN IP-2B

The Furman IP-2B Iso-Patch ($99) is a small, stomp-box-size unit that is designed to eliminate the buzz and hum caused by ground loops. Such a tool is useful to have on hand for on-the-spot fixes in any live or studio situation.

Sound systems typically have multiple paths to ground through AC cables, metal racks, and audio-cable connections between multiple units, such as mixers and effects processors. The IP-2B provides transformers that isolate the audio signal, passing it through a magnetic coupling and thereby breaking ground loops to remove hum. These transformers feature parallel windings and high-quality magnetic cores for ample transient response and minimum low-frequency distortion.

The IP-2B has two channels, each with XLR and ¼-inch TRS inputs and outputs, providing balanced/unbalanced conversion. Its frequency response is rated at 10 Hz to 200 kHz (+0/-0.5 dB @ -10 dBu) with no more than 0.5% total harmonic distortion from 20 Hz to 20 kHz at 0 dBu. Furman Sound; tel. (707) 763-1010; fax (707) 763-1310; e-mail info@furmansound.com; Web www.furmansound.com.

Circle #414 on Reader Service Card

SOUND ADVICE ▲ ▲ ▲ ▲

▼ SOUND DOGS

Some sound-effects CDs have a long list of sounds but only a few that you are interested in using. Now you can download individual sound files from Sound Dogs, a Los Angeles–area sound-design team that works on feature films. A portion of Sound Dogs' sound-effects library is posted on its Web site, where files can be purchased for a nominal fee that varies depending on length, bit rate, and sample rate. (Prices start at 50 cents.) The purchaser may then use these files royalty free.

Sound Dogs offers more than 10,000 sound files in several formats, including AIFF, AU, and WAV. The files are organized on the Web site into many categories, from babies and buses to machines and monsters to winds and witchcraft. Each file is presented with a short description of the sound, where and when it was recorded, and its length. Sample rates of 11, 22.05, 44.1, and 48 kHz are available, as well as resolutions of 8, 16, and 24 bits.

When you shop at Sound Dogs' Web site, you can audition each sound you are interested in before you tag the ones you want to purchase. You have the option of specifying a segment of a file to purchase—for instance, you could buy only 10 seconds of a 20-second ambient sound file—which can save you money. Sound Dogs; tel. (310) 244-7988; fax (310) 244-5422; e-mail sounddogs@sounddogs.com; Web www.sounddogs.com.

Circle #415 on Reader Service Card

MARCATI DISTRIBUTION

Miroslav Vitous Keyboard Series, a new line of sample CDs from Marcati Distribution, features samples conceived and recorded by Miroslav Vitous, a founding member of the jazz-fusion group Weather Report and creator of the Symphonic Orchestra Samples CD library. The series' first offering, *New Instruments* (CD-ROM; $699), is available in Akai S1000, Digidesign SampleCell, E-mu E4, Kurzweil K2000/K2500, NemeSys GigaSampler, and Roland S-700 formats. It offers approximately 100 separate instruments and 60 combinations of acoustic sound sources. The sonic blends on this disc are meant for use in many styles of music, including rock, pop, R&B, soul, jazz, classical, and world music.

The sound sources used for the disc are not necessarily musical instruments. The samples were edited on a computer using Digidesign SampleCell and *Sound Designer*, Steinberg *WaveLab*, and BIAS *Peak* to sound like familiar instruments with unfamiliar overtones, timbres, and harmonics. Marcati describes the resulting samples as "new instruments" for synthesizers. Individual samples resemble instruments such as acoustic piano, electric piano, synth pads, Clavinet, steel drum, accordion, and sitar, among others. According to Vitous, the samples have a natural, acoustic quality that is bolstered by plenty of low frequencies.

A demo disc and a booklet containing descriptions of each sound are available for free from Marcati. Marcati Distribution; tel. and fax (203) 323-8300; e-mail mvsos@discovernet.net; Web www.marcati.com.

Circle #416 on Reader Service Card

▼ SOUND SUBSCRIPTIONS

Shane Etter's Sound Subscriptions ($50/year) is an Internet-based service offering new program sounds every month for Kurzweil's K2000 and K2500 synthesizers. Bored with what

he felt to be lackluster preset sounds, Etter created Sound Subscriptions as a way for K2000/K2500 users to broaden their sound palettes and get inspired. All sounds were created from scratch using the 31 synthesis algorithms available within Kurzweil's V.A.S.T. digital signal processing engine.

On the first of every month during your subscription period, you receive e-mail containing a bank of 100 new sounds for use in dance, industrial, hip-hop, ambient, and other styles. Each batch includes electronic and synthetic timbres, as well as new drum-kit programs. Subscribers simply save the patches to a PC-format floppy disk and load them into the Kurzweil synthesizer.

A demo containing free sounds can be downloaded from the Sound Subscriptions Web site. Also available is *Dr. Mach* ($75), a CD-ROM containing 40 different drum-machine kits, with more than 1,000 individual samples that have been keymapped for the K2000/K2500. Sound Subscriptions; tel. (303) 604-9318; e-mail soundsubs@ aol.com; Web members.aol.com/soundsubs/soundsubs.html. ❀

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WRH

TECH PAGE

Quantum Sound

A new discovery proves that sound is quantized.

By Scott Wilkinson

One of the hallmarks of 20th-century physics is the development of quantum mechanics, which describes the properties of matter and energy at very small scales. At its heart, this highly abstruse and esoteric branch of physics is actually quite simple: all matter and energy ultimately consist of discrete units called *quanta*.

Sound has always appeared to be one of the rare exceptions to quantum mechanics; it seems to be a purely analog form of energy. Recently, however, Dr. Leonardo Da Capo of the Bologna Sound Harmonics Institute of Technology (www.bshit.edu) made a titanic discovery: sound actually consists of tiny, discrete quanta, which he calls *phonons*. These phonons are individual, separate packets of energy, much like photons of light.

When asked to describe his revolutionary discovery, Da Capo exclaimed, "Let me go back to the beginning." The original idea was inspired by the harmonic series, which is itself quantized into discrete steps. He went on to explain that when phonons interact, they produce difference tones that we perceive as audible sound. The frequencies of individual phonons are expressed in *yottahertz* (septillions of cycles per second).

Phonons were first detected using a sonic cyclotron (see **Fig. 1**), which will soon be used during half-time shows at Seattle SuperSonics basketball games. Later, Da Capo identified an even smaller type of phonon that he calls a *phonino* or *microphonon*. Like quarks in subatomic particles, several phoninos are bound together to form a phonon. In addition, phonons can combine to create sonic molecules called *polyphonons*, which give rise to timbre.

Phonons share another trait with subatomic particles: antiparticles with equal but opposite attributes. In the sonic microcosm, these are called *antiphonons*. They are the opposite of phonons in that their frequency is very low because they are somewhat like aliasing artifacts. There seems to be a sort of cosmic Nyquist frequency that is twice the lowest phonon frequency. (This relates to the sampling frequency of the entire universe, but that's a subject for another column.) As a result, antiphonons appear at frequencies very near zero. Of course, we can't hear these antiphonons, but Da Capo suspects they might be the cause of unexplained bumps in the night.

Da Capo is now working on ways to transmit phonons over standard modems. This could ease the bandwidth requirements of streaming audio, because the essence of sound is sent instead of the entire result. The final sound would be reconstructed at the receiving end. Of course, the first-generation prototype of such a codec is called a *telephonon*.

Unfortunately, a Bolognese bureaucrat by the name of Enrico Fermata recently put a hold on the project. He believes that Da Capo is a psycho acoustician who must be stopped before he does permanent damage to the surrounding buildings with his booming sonic cyclotron. Fermata points out a nearby garden, where several pumpkins have been smashed by resonant antiphonon radiation leaking from the device.

However, Da Capo remains convinced that his work will lead to some major breakthroughs in the new field of quantum acoustics. He has even had T-shirts made with the slogan "Hooked on Phonons," which he is selling to raise independent funds for his research. If Da Capo can cut through the governmental cacophony surrounding the project, his discovery could rewrite the physics textbooks and earn him the Nobel Peace Prize. In fact, he's already written his acceptance speech, which starts, "Let me go back to the beginning...." ●

FIG. 1: In a sonic cyclotron, the music goes 'round and 'round and 'round, and it comes out here.

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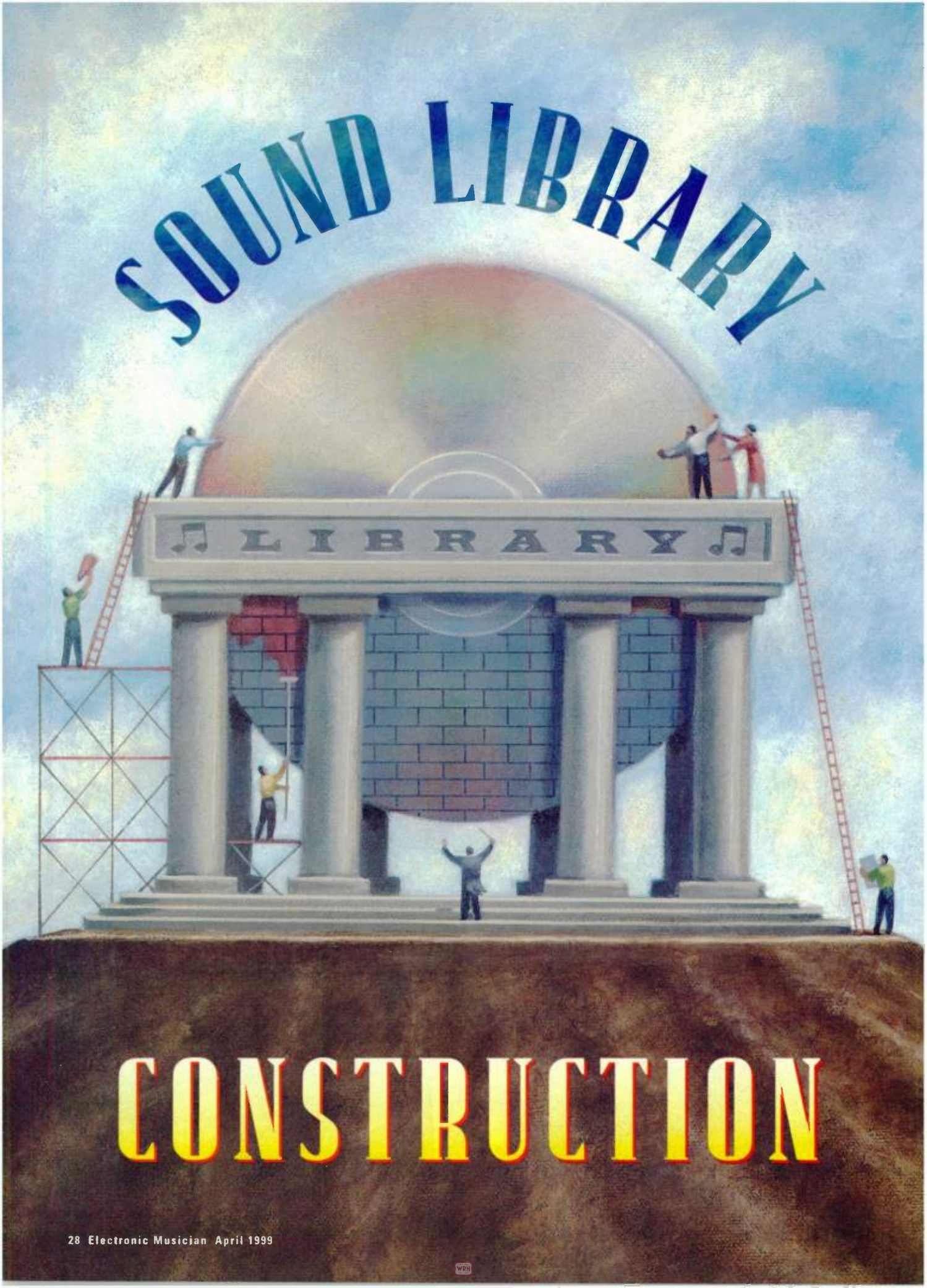

SOUND LIBRARY
LIBRARY
CONSTRUCTION

Create your own sample library for fun and profit. Let the pros show you how.

By Gino Robair

Sound libraries, such as sample CDs, are ubiquitous in the personal or commercial studio. If you have ever used them, you may have wondered how they're put together. Perhaps you have amassed an archive of recordings or instrument-specific patches and have thought about compiling your own sample collection, if you only knew where to begin.

Armed with the items typically found in a personal studio (DAT machine, DAW, CD burner, sampler), you can create a professional-sounding library to meet your needs. All it takes is some vision, planning, inspiration, and a good deal of editing.

I surveyed a handful of successful professional sound designers, manufacturers, and distributors about the various aspects involved in creating a sound library. Whether you're recording breakbeats, exotic instruments, production music, sound effects, or any combination thereof, the following information will help you maximize your time and energy as you organize your favorite sounds for personal use or commercial distribution.

FROM THE TOP DOWN

The first order of business in making a quality sound library is to do some preproduction planning. Begin by determining your goals for the project: Is it for personal use or will you be sharing it with others? What are the intended uses for these sounds? What format will work best? Do you plan to license it to distributors? Setting a goal for your project will help you answer many of the questions that will arise as you get further into it.

For example, your plan might be to create and license a library in a specific sampler format to an established company. In this case, you'll need to have the most recent update of the machines that read the format you've chosen, with the maximum amount of RAM available. This will allow you to take full advantage of every nuance that machine has to offer.

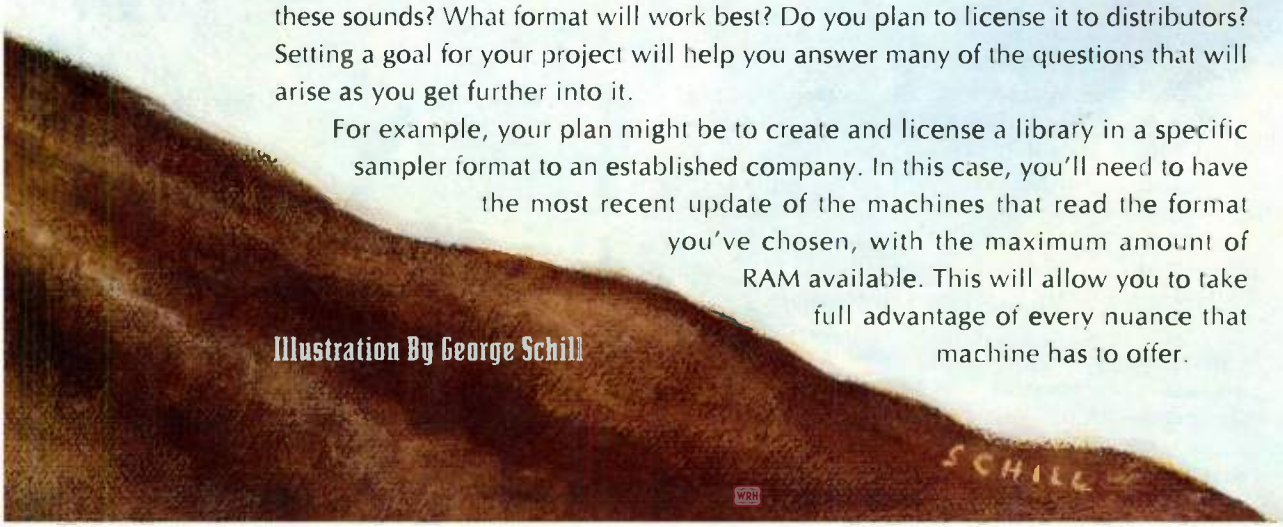
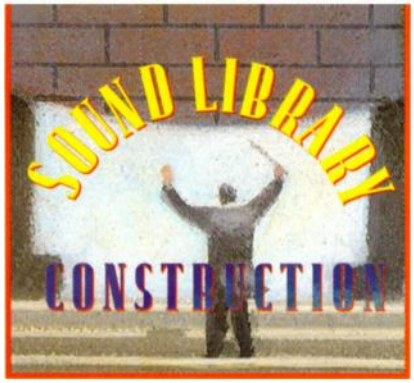


Illustration By George Schill



Perhaps your objective is simply to create an audio CD for your own post-production sound-design purposes, in which case you'll want to concentrate on your recording and CD-burning setup.

Remember that you can always modify any goals that you set for yourself as the project progresses. After all, you may find yourself heading down a new path once you've gotten started.

ORGANIZATION

Anyone who has used a sound library will tell you that a well-organized disc is worth the money it costs to create or buy it. Determining the best way to organize a collection of sounds depends on how the library will be used, as well as the taste and style of the user.

Sound designer Jeff Darby assembled the 25-disc sound-effects library for Earwax, a production company in San Francisco. The discs are housed in a large three-ring binder with a log that breaks down each disc into specific categories and subcategories.

"The Earwax library is made up of Sound Designer II files, primarily. Each CD-R is a data disc containing Mac files and folders, burned with Adaptec *Toast*," he explains. "If you're using Pro Tools, you import the files directly into the application. This method eliminates a couple of steps: you don't have to edit, normalize, or digitize sounds each time. Once they're done, they're done.

"The Earwax library is mainly a by-product of jobs we've done," he continues. "We then organize the sounds from the jobs into categories such as Whooshes, Machines and Tools, and Weapons and Violence. In the Whooshes folder, you'll find a folder of swishes and perhaps one of dopplers. Weapons and Violence contains subcategories such as Hits and Thuds. The Thud folder may, in turn, contain 45 different thuds. Usually, there are a lot of subcategories."

Organizing sounds into categories is a subjective art and is often a matter of personal preference. "In the Ambience folder, for example, you might have nature, city, or bird ambiences; I wouldn't necessarily put a bird ambience in an animal folder. There are a lot of gray areas. We usually build the library according to the way we think and what our needs are."

Earwax has considered putting together a "best of" collection for commercial release. "At the rate it's going," says Darby, "we'll probably have another ten discs in the library by the end of the year, which will give us plenty to choose from." Their biggest challenge will be categorizing the material in such a way that it will be intuitive for a new user.

Sounds are organized in libraries in a number of ways. Here are some of the more common ones.

Instrument. If the project features a collection of instruments, grouping each instrument or class of instruments together makes good sense.

Key. Compiling sounds by key is an especially useful way to organize a library of instrument-specific riffs and licks (see **Fig. 1**).

Tempo. If the disc will be used in a dance or club environment, having the samples arranged by tempo in bpm (beats per minute) is essential. An elegantly designed library will first give you the entire groove (full orchestration) at a given tempo, followed by each component of the groove, individually.

Theme or genre. This does not necessarily mean a "musical" theme. If the library is a sound-design collection for post-production, organizing by theme (for example, sci-fi, western, and ethnic) would be a logical way of working. If you are assembling a library of music cues, you could group tracks by genre (classical, rock, jazz).

Effects type. This category involves classifying effects by how they sound (for example, pops, applause, and gunshots) or what their functions are (ambiences, hits, effects).

Combinations. For a comprehensive collection of samples, you may need to combine some of the above organizations. For example, you could divide one portion of an orchestral disc into instrumental sections (say, brass and woodwinds); another into orchestration (groups or soloists); and so on, on down to individual notes and effects. Another section of the disc could be organized into tempo- or key-related selections.

DOCUMENTATION

A user-friendly library relies on full documentation. A great-sounding disc becomes difficult to use if the liner notes are cryptic or nonexistent. If you are planning to license your library, thorough documentation is a must and begins the moment you press Record.

Doug Morton of Q Up Arts recommends keeping detailed notes throughout the recording session. "Make sure you document what you're doing so that everything will be clear to the person who works on the material later on."

Your recording notes will be useful as you edit and loop your samples. Notes on mic choices, mic position, recording location, and any processing that occurred, are handy when you need to combine different parts of a sample or create Velocity layers. Such details may seem mundane at the time, but they may save you from having to rerecord elements later.

Disk One:
Jazz Loops

Track 01
Demo

Track 02

1	120	E Minor
2	120	E Minor
3	120	E Minor
4	120	E Minor
5	120	E Minor

Track 03

1	98	E Minor
2	119	E Minor
3	120	E Minor
4	120	E Minor
5	121	E Minor

Track 04

1	120	E Minor
2	120	E Minor
3	120	E Minor
4	120	E Minor
5	120	E Minor

Track 05

1	120	E Minor
2	096	E Minor
3	096	E Minor
4	096	E Minor

Track 06

1	135	E Minor
2	135	E Minor
3	086	E Minor
4	086	E Minor
5	120	E Minor

Track 07

1	120	E Minor
2	120	E Minor
3	120	E Minor
4	120	E Minor
5	120	E Minor

Track 08

1	99	A Minor
2	120	A Minor
3	120	A Minor
4	120	A Minor
5	113	A Minor

Track 09

1	113	A Minor
2	113	A Minor
3	113	A Minor
4	113	A Minor
5	120	A Minor

Track 10

1	108	A Minor
2	120	A Minor

COURTESY EYE & I PRODUCTIONS

FIG. 1: This is a page from Voice Crystal's audio CD *Song Starters: Jazz Loops and Chords*, vol. 1, by Frank Gambale. Notice how the loops are organized by bpm and key. Elsewhere on the disc, the tracks are organized by key and chord type.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Vector Automation
Real-time
MIDI Plug-ins
Sync
AVI
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DR-1

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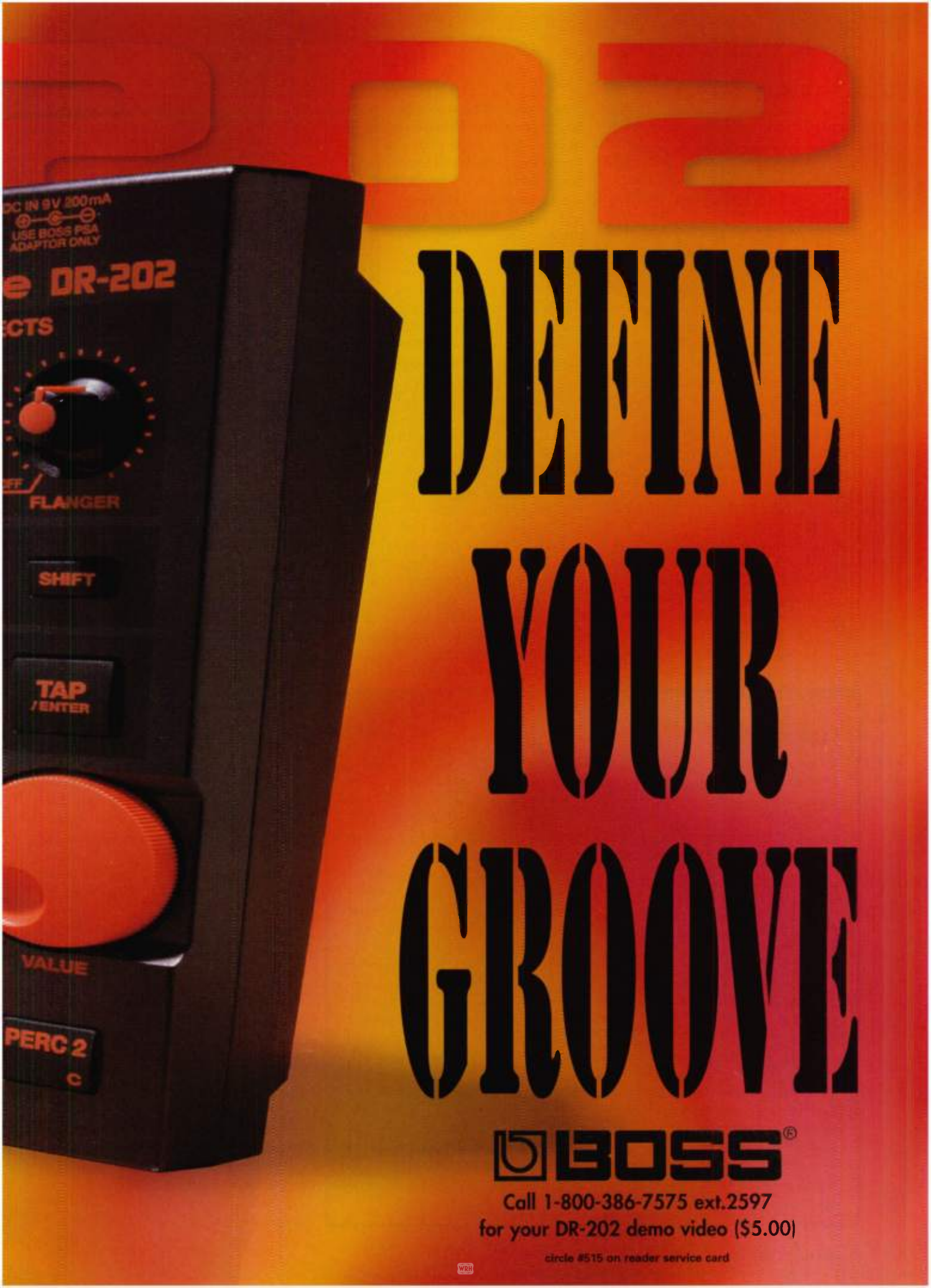
- Realtime sound-shaping control over Tempo, Filters, Low Boost, Mutes, Effects, and Drum Rolls.
- Includes essential TR-808/909, TB-303 sounds as well as many others for a total of 128 rhythm kits.
- Groove-friendly 3-track sequencer designed to let you manipulate onboard patterns or build your own beats and basslines from scratch.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Your session notes will also be helpful when you compile your liner notes. The kind of documentation you publish with your library will depend on the format you're working in, as well as its intended use. Notes for an audio CD of loops and grooves should include the name and number of each track, tempo, number of bars in each groove, start times, track length, key, and instrumentation. When the groove is broken down into individual components, these should be noted as clearly as possible.

For a CD-ROM release, it is crucial to include format-specific information in your documentation. A library in the Akai format, for example, should give the location, volume, program name, sample size, and length of bars of each sample. In addition, you might include key, tempo, crossfading, and Velocity-switching information, as well as whether the sample is stereo or mono. Providing details about each instrument or voice that is sampled is also useful.

Spectrasonics' *Symphony of Voices* CD-ROM is a four-disc set that comes with two elegantly produced, 50-page booklets. Booklet 1 is a CD-ROM directory that includes setup instructions, system recommendations, and a list of patches and performances. Every sample is listed with key and tempo notation. The liner-note booklet includes biographies of the artists, performance tips, an overview of the organization of the sounds, descriptions of the recording and editing equipment used, the philosophy behind the project, and a list of frequently asked questions about using the company's sound libraries (see **Fig. 2**). Such a cornucopia of documentation enhances the users' knowledge about the sounds as well as the pleasure they get from using the library.

ROB FORLIN

FIG. 2: Proving that you can never have too much information, Spectrasonics' *Symphony of Voices* contains four CD-ROMs and two highly informative booklets.

BRAIN STORM

When coming up with ideas for projects, Eric Persing, creative director at Spectrasonics, begins by imagining what he would like to hear in a sample library. "I might begin by thinking about what's not in our catalog, or by considering what people are interested in." Product-registration cards give him a sense of what customers are looking for in new products.

"Other times it's a whim," he continues. "Sometimes we've started with just the title. *Distorted Reality*, our most popular release, began that way. We also have to consider what we are capable doing. Thinking things through is essential because [creating a sound library] is a long process. Several of our libraries have taken up to two years to complete."

Morton says that at Q Up Arts, quality and uniqueness are what drive new product development. "We try to find something that's different, or find something that hasn't been done well and do it better. We also try to avoid stylistic trends. Our niche is world/ethnic/esoteric content, such as *Voices of Native America*. It's the harder-to-find kinds of things that we prefer."

FORMAT ROULETTE

Whether or not you're interested in going commercial with your sounds, a brief discussion of how companies choose a format for their sound library is useful—especially if you're planning

LICENSING

When you purchase a sound library, you're not buying the sounds any more than you're buying Peter Gabriel's songs when you buy his record. The sounds on a sample disc are licensed to you for use in *your* sound productions only. Each manufacturer handles licensing differently, so it's important that you read the enclosed "licensing agreement" for every collection you purchase.

Typically, the license allows you to use the sounds for your own productions exclusively. Unauthorized use of the collection includes lending, renting, reselling, copying, dubbing, and reconfiguring the samples for resale.

Companies often approach each kind of use of the sounds separately. Record production, film and television, and multimedia projects are each treated in a unique way. Some uses may require a separate agreement with the publisher or may be forbidden entirely. Sometimes you're asked to credit the sound library, depending on the usage.

Why are sound-library publishers so particular about licensing issues? Consider that each company has spent a lot of money hiring musicians for the recordings and invested countless hours in editing, looping, and programming each sound. The people who do this expect to get paid for their work, just as you and I do. In addition, many companies guarantee that the sounds they've licensed to you are copyright clean, so you won't have legal problems down the line. That aspect of a sound library is an important one that's often overlooked: for a relatively small fee, these companies are providing you with the tools to continue creating music (and making money) in perpetuity. All they ask in return is that you abide by their licensing agreement.

to purchase any gear in the near future.

There's a wide range of formats to consider, the most common of which are Akai, E-mu, Ensoniq, Kurzweil, Roland, SampleCell, audio, WAV, and AIFF. For the publisher, each format has its strengths and weaknesses that extend beyond the capabilities of the machines themselves. For example, publishers must gauge how popular the formats are in the marketplace, where the majority of their sales will be, what kind of library they're creating, and who will be using it. A number of manufacturers I spoke with agreed that the audio format was the best seller but had the worst profit margin.

"In times past, we did audio discs first and worked toward Mac AIFF," notes Paul Korntheuer of Rarefaction. "Even though CD-ROMs are superior, people are used to sampling from audio discs."

Melissa Reuther of Time + Space agrees. "For us, everything comes out in audio first. The CD-ROMs mostly come out in Akai format first, because that format is the leader in the U.K. market. However, the mainstream buyer is buying audio discs."

Q Up Arts' Morton says that "certain kinds of sounds are good for certain markets. Generally, we sell big in Japan, and Akai is the king there. We also do CD-ROMs for SampleCell, because we develop in SampleCell. And then, of course, there's Roland, Kurzweil, and audio."

"If you're going to make a business out of it, you have to support all formats," says Spectrasonics' Persing. He says that releasing multiple formats simultaneously is one of the company's recipes for success. "The market is different for each format, so we do different quantities of each. At the moment, Akai is the most popular format on the market. Roland is a good format in the United States because Roland is the only company that has made a serious library of its own. Just because lots of units of a keyboard are sold, it doesn't mean there will be a CD-ROM market for it."

It's no surprise that Persing, as chief sound designer for Roland since 1984, prefers to work in the Roland format. "We will begin a new release in the Roland format, then we'll do an Akai version. We make our audio CDs from the Roland format, which is the opposite of what other companies do. And because of size limitations, the audio CDs have about half the material that the CD-ROMs have," he explains.

THE RECORDING SESSION

There are a number of things to consider when making recordings that are destined for use in a sound library. First, you must determine the recording chain, from the microphone to the storage format. These variables are often determined by economics. Perhaps you'll be able to borrow equipment. If not, what kind of gear can you afford to buy? If purchasing it isn't possible, how much can you spend on renting it?

Despite the current trend toward higher bit and sampling rates, many of the sound designers I interviewed still record to DAT and work in 16-bit resolution. The common explanation was that samplers and compact disc players are still primarily 16-bit machines.

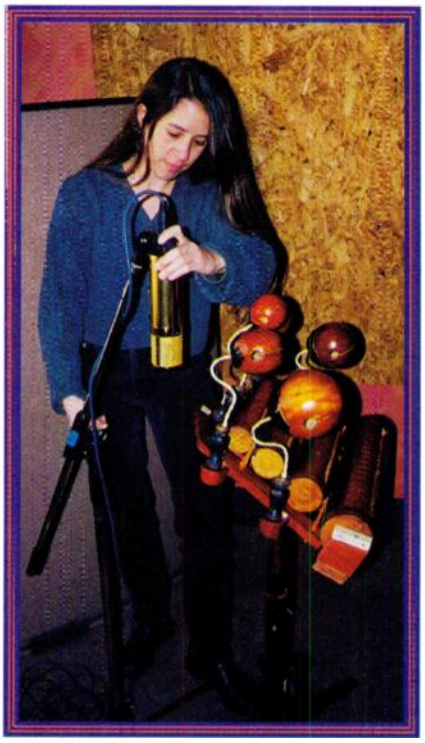
Of course, you want your master tapes to be in the best-sounding, most robust format possible. If you're recording in a studio environment, you have higher resolution options, such as the 20-bit ADAT XT20 and LX20 from Alesis, Tascam's DA-45 HR 24-bit DAT machine, hard-disk recording, or high-quality analog tape. Out in the field, however, the most affordable format is still the portable 16-bit DAT recorder.

For location recording, sound designer Jim Miller uses a portable setup that includes AKG C 414 and Earthworks TC30K mics, and an Oram MWS preamp going directly to a DAT recorder.

Darby's field rig includes a portable DAT with a Lunatec V2 portable preamp by Grace Design and a Sennheiser MKH 816 shotgun mic. Because much of his work is for film and television post-production, this setup is portable enough to allow him to get into tight places when necessary.

"The recording format depends on the application," says Ilio's Mark Hiskey. "Sometimes we go to 20-bit ADAT or 24-bit. Sometimes we go straight to DAT at 44.1 to avoid sample-rate conversion. We're considering 96 kHz and anticipating DVD as a viable delivery system. The only question is when."

Daniel Fisher, veteran Kurzweil programmer and director of soundware engineering at Sweetwater Sound, records simple instruments straight to DAT. "For things requiring lots of mics, I'll use one of the 20-bit ADATs and four different stereo mic combinations."



ROB FORLIN

FIG. 3: Karen Stackpole prepares for a sampling session at Headless Buddha Labs. Many sound libraries are recorded and edited by outside contractors working in their own studios.

Morton does everything on the Mac. "We record directly into Pro Tools/24. Before that we were working in 20-bit for about a year."

Don't think that you should put off recording just because you don't have state-of-the-art gear: ingenuity and recording skill can overcome many deficiencies in your setup. Besides, there are other considerations that will raise the quality and usefulness of your samples.

The majority of the sound designers that I asked record their material flat, preferring to get the best sound possible by using the shortest route between mic and recorder. Mic placement and room acoustics, therefore, play a major role during the initial sessions.

"I prefer a nice, quiet studio with no reflections. Many budget restraints, however, require that you get your recordings where you can," says Miller. "I wouldn't add EQ while recording unless there was something extremely wrong with the sound. I prefer to do that later."

I asked a number of sound designers whether tuning was an issue when recording and whether they used tuners during their sessions. Morton says that Q Up Arts always stresses tuning. "This

is the sort of detail that should happen during the original session."

Others took a more laissez-faire approach to tuning. "It's never been a problem," says Miller. "Usually the players will bring a tuner themselves. However, it's simple to correct tuning in the Kurzweil. You just save the tuning as part of the sample itself."

"But," adds Fisher, "the samples must have exactly the same tuning. Otherwise, when you add reverb and switch Velocity levels, you will hear chorusing in the reverb."

Fisher says that, if there's a problem, he prefers to "EQ the noise out on the Kurzweil. With the K2500, you can work on a sample and then resample it digitally using V.A.S.T." He cautions that you have to listen carefully during the recording session. "You have to be insane about noise levels. Pay close attention to ambient room and electrical noises. Remember that when you play multiple notes of these samples, you're adding the room sound and RFI each time you add a note."

A final consideration during the recording stage is getting permission to use these performances. Unless you play every instrument yourself, you will need to get written permission from the players to use their sounds in a commercial project (whether on a CD-ROM or in a production). If you intend to license or sell the collection, having copyright-clean samples is a must. Think about this before you contact musicians for the recording session. Be sure to have an attorney-approved contract stating your intentions, and make sure the musicians understand and sign the contract before the session is booked.

FOCUS

It's easy to spend hours setting up mics and getting the right sound. However, it's important to conserve your mental and physical stamina for the recording session itself.

For Fisher, preserving attention span is important. "People will spend all day on the miking and sound. But after an hour of recording, you'll find them reading a magazine rather than paying attention. It's very easy to have entire ranges of an instrument get wrecked because you lost your concentration for a period of the recording session."

Attention span is a two-way street. "Be willing to pay your performers," Fisher adds. "Tell them that this'll be the hardest and most boring—but most exacting—performance of their life. Let them know that this is a major deal."

And be thorough in your search for the most useful sounds. The more prepared you are before a session, the smoother it will run and the more you will accomplish.

Before doing a sampling session with the ROVA Saxophone Quartet, sound designer Thomas Dimuzio made a wish list of things that he thought would be useful in a sax-quartet library. "I brought along a list of ambiences, concepts, and feels that I wanted to get on tape. It turns out that ROVA brought an almost identical list. Because of this, we recorded more than enough material for a sound library."

DETAILS, DETAILS

"Get lots of Velocities," suggests Fisher, "so that you have lots of choices later. But don't get too literal about things when programming: if an E sounds better as a D, put it there. Also, remember that the stereo position of two or more samples has to be perfect, because the ear can hear stereo location shifts."

Fisher is also an advocate of keeping the sampler set up nearby to see how the samples work side by side on the keyboard. "Having the sampler nearby is useful for getting the sounds to blend all the way up the scale."

"One of the most important things you can do during the original recording is pop the sounds into a keymap and see if they work on the instrument and in scales. Many times, you won't know that the sounds don't work until you put them all side by side. Very few people do this. However, this keeps you from getting gaps where the sound isn't right. Sometimes you'll find that you need to get a better recording of a single note to even out the scale.

"After you've built your keymap, don't be afraid to throw out a note that doesn't work and stretch the others to cover it." Fisher adds.

Miller agrees that having an instrument nearby is a good practice. When he is creating a project destined for the Kurzweil format, he loads the recordings into his K2500 and tries a little programming to get a feel for how the samples will sound in performance. "I like each sample to have a slightly different sound across the keyboard map. When done well, this will give the instrument more of a real playing feel."

Fisher believes that the more you know about how an instrument sounds,

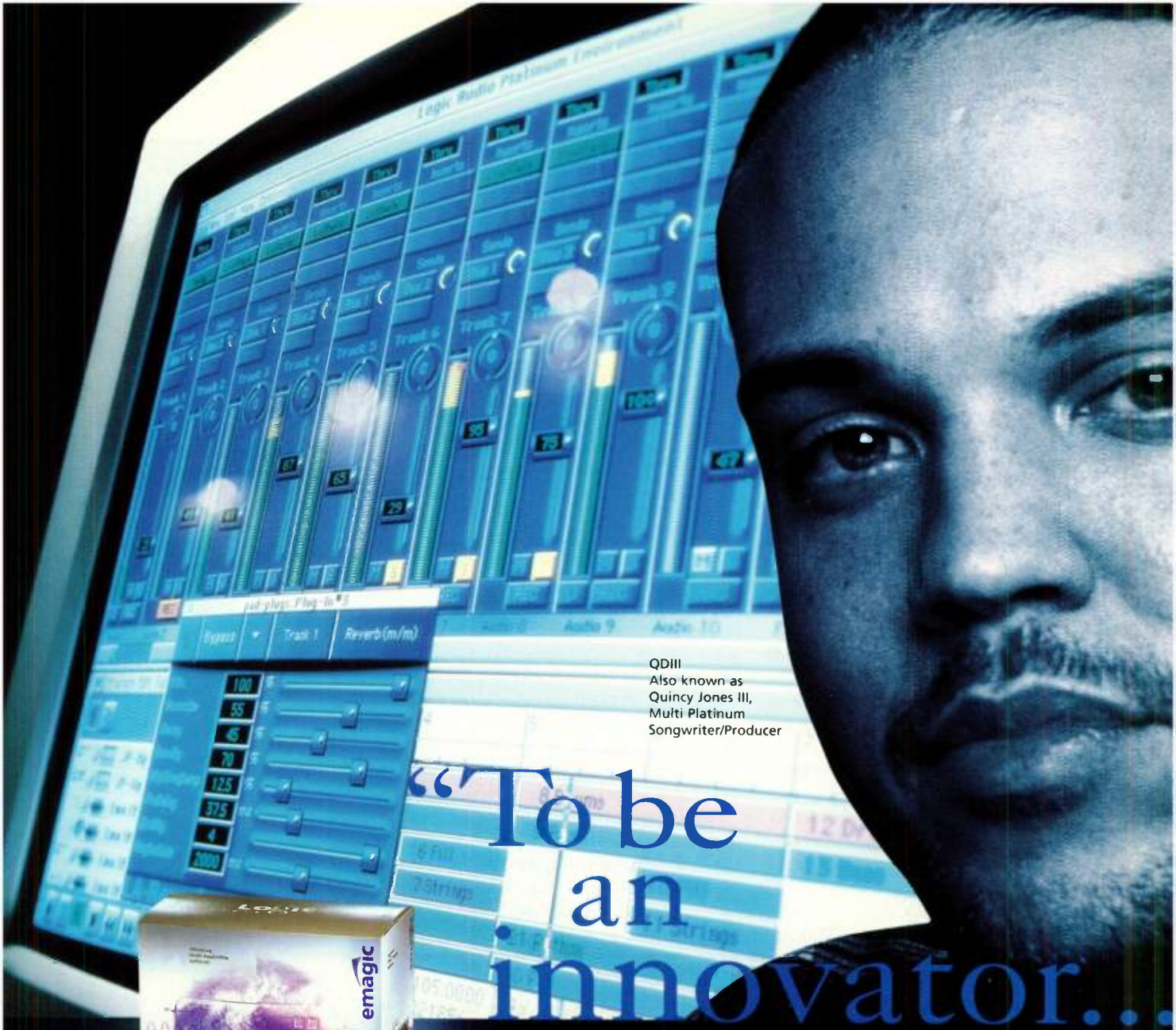
PIRACY

Every company that I spoke with said that piracy of samples is a major issue worldwide. One company said that the sample-CD industry would be at least ten times bigger than it is now if it weren't for piracy. That means that the development process of new and unique sample collections has been seriously hampered because of the unauthorized sale and distribution of samples.

It's certainly fair for musicians to upload their own custom patches or samples to a Web site. It's another thing to upload something you've purchased from a publisher, whether in its original form or modified. We're talking basic intellectual property rights.

Make sure that you're not supporting unscrupulous activity when you download sounds from users' groups or other Internet sources. Any unauthorized reproduction of the sounds, whether posted on a Web site for distribution or copied to DAT for use by persons other than the original user, is harmful to the company that produced the project.

It's important that you're aware of piracy issues and respect the work of the manufacturers just as you would like people to respect your own musical creations. If it weren't for these companies taking financial risks, we wouldn't have so many fantastic releases in the first place.



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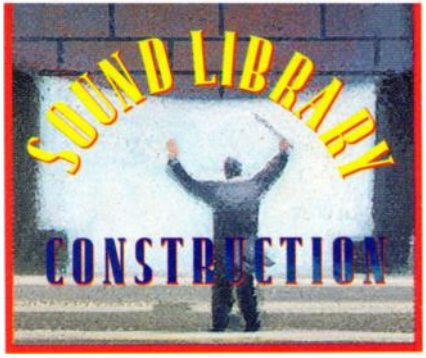


the easier it will be to create a reproduction for the sampler. "When the session's over, have the musician just play, so you can hear what a real performance sounds like. It'll give you ideas on how to structure your sample programming. Try to reproduce that performance with the samples."

Through experience, the professionals also know that *how* the sounds are used in a library often determines the best way to record and program them.

"Sometimes you need to predict what the sample's going to sound like in the end; it may not sound the way it sounds in real life." Fisher points to the acoustic guitar as an example. "Recording the guitar's open strings and putting open-string samples on each key, rather than getting the individual quirks of each fret, gives the sample a more natural sound. Sometimes a vanilla recording of an instrument is better than one with lots of different clicks and buzzes."

EDITING AND LOOPING

Now the real work begins. Making many detailed samples from a collection of source recordings is a major task. Whether you are using a looping program, such as Antares's *Infinity*, or working in a more generalized digital audio editor, the biggest challenge to the sound designer is seamless editing and looping.

Hiskey says Ilio Entertainment's first priority is to achieve high-quality sound. "We can't afford to put out a product that has clicks in the loops."

Like many Kurzweil programmers, Miller says that he does his work primarily in the K2500 itself. "The way I work now, I go from DAT to Mac, using BIAS *Peak* to slice it up, and then send it where it's going to go, which, for many projects, is the K2500."

Earwax's Darby tries to keep as many of the samples intact as he can. "Often, they're from previous jobs, so they've been trimmed. Because we use Pro Tools, it's not a problem to adjust the sound to fit the moment."

"There are compromises no matter what you do," explains Dimuzio. "It all depends on the source material." On Rarefaction's *Etymology*, a sample library of guitar and cello sounds played by Fred Frith and Tom Cora, Dimuzio often used crossfade loops. "But some folks may not like crossfades, so I also included chronological loops, where I went in and cut loops from the beginning of the sample to its end. When you place them side by side, you get the entire sample. Otherwise you have its component parts."

Many elements come into play when you begin creating individual samples. Sound quality, tuning integrity, and level are all things you must consider when choosing the best take. Normalization is an aspect of sample design that Morton thinks important. "Make sure all the levels are hot so that sampler output is optimized. Also, watch that stuff doesn't get truncated too short. Give the instruments their full decay before chopping them off."

Fisher suggests that if note attacks don't match, you should copy and paste the attack transient you like to the one sample that has the weak attack transient. "And make sure your loops don't change pitch," he warns. "Your ear is most sensitive to pitch, especially with shorter loops. Moving the end of the loop point will make the difference. Don't just listen to the sample dry: listen to it in reverb, because, again, you will hear chorusing when both pitches appear together in the reverb."

Fisher also knows that the ear is good at perceiving when a short loop begins. "If you have a RAM limitation, you're better off having fewer samples with

ROB FORLIN

FIG. 4: For a project as comprehensive as Rarefaction's *A Poke in the Ear with a Sharp Stick*, 7 to 10 GB of recorded material go into every 650 MB disc.

WHAT YOU CAN EXPECT IN SALES

Despite the apparent glamour of the sample-CD biz, the companies that publish libraries are not the huge megacorporations that you may think they are. They're usually small businesses, started by sound designers who thought that others might be interested in their work. While some publishers do much of the work in-house, others commission and license projects from sound designers like you and me.

Before you begin seeing big dollar signs, however, let's begin with a reality check. Compared with the number of units that traditional music labels sell, sample-CD sales (whether they're audio discs or CD-ROMs) are extremely modest. Depending on the title, format, style, and price, a sample CD might be considered a success if it sells from 750 to 1,000 units, but the average release sells below that figure.

Selling 600 copies of a CD-ROM priced at $199 may not sound that bad, considering how cheap it is to mass-produce CDs these days. However, when you consider that creating a high-quality library can take up to two years, you begin to see that there are a few challenges in making back your initial investment.

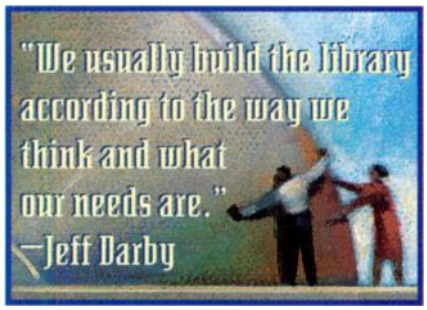
Style-oriented releases (for example, current dance flavor) generally have a shorter marketing life span than their more esoteric counterparts (such as a harpsichord collection). Some sample libraries continue to sell well for manufacturers, sometimes into the thousands, and this kind of success helps the manufacturer invest in new releases to meet market demands.

longer loops than the other way around. Especially with something as complex as a multistring instrument like the piano: you need loops of two to three seconds or more. They should be as long as you can get away with."

WILL WORK FOR SAMPLES

Many sound-library publishers don't have in-house recording and editing facilities. That fact means that much of the work on a project is performed by outside contractors working in their own studios (see **Fig. 3**).

Most of the companies I spoke to expressed interest in hearing about new ideas for sound libraries and collections, especially ones that may already be completely implemented into one of the popular sampler formats. As the



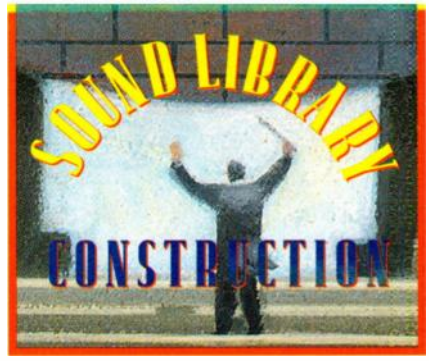
market continues to grow, there is also a need for skilled freelance editors and programmers. Because of the amount of time it takes to program a set of samples, you can imagine that many companies have a difficult time keeping up with the demand for new sounds.

"There's quite a bit of work to be had converting libraries for companies," explains Spectrasonics' Persing. "The best part of the project is recording the musicians—making the raw recordings. The next part that's fun is when it's done. All the other parts are technical and tedious because of the amount of organization and documentation that's required. Just coming up with names, it's hard not to repeat yourself. A lot of thought goes into these details."

Persing says that to be a good programmer, you need to have full knowledge of what he calls the "arcane archives" of each machine. "If you're going to make something for the Akai S1000-series filters, you need to know that they respond differently from other Akai versions. Filters are standardized in Roland and Kurzweil machines, but not in the Akai instruments. So, for a certain project it might be necessary to leave the filters alone in that format. You have to be aware of details like how the envelope times work on different models. The more musical you want to make it, the more risky it is. But the rewards are that much greater."

To Persing, the creativity that goes into the details matters. "Controller mapping makes a big difference," he says. "If you want to have a filter sweep, a mod wheel or data slider can add in more or less of the effect. That way, folks can tailor a sample in real time."

Fisher points out the "paradox in creating multi-Velocity-strike samples. You want two different Velocities, but when you switch between samples, the timbre is different. People complain about that. What you have to do is make the crossover points seem invisible. You have to have a separate programming layer for each Velocity range: as you get to the end of one sample range, its brightness should match the darkness of the new layer."



Fisher also notes that there are reasons to do cross-switching rather than crossfading. "Crossfading rarely works on natural instruments when you're doing multistrikes. While they're crossfading, you're hearing twice as many strings in that crossfade window. With switching, when the Velocity is 0 to 60 you're hearing the low sample; 61 to 127 you're hearing the other. Also, when you crossfade, you're eating twice as much polyphony during the fade."

But achieving utter realism is not the only measure of success with a sample, says Fisher. "Many instruments that decay do so far faster than your brain thinks they do. So you have to exaggerate the decay time on certain instruments—such as piano, guitar, and bass—so users won't complain about the sounds being 'stubby.' They want a far more linear curve that hyperexagerrates the decay time of those instruments, which is why compression is used so often on those instruments in the recording studio."

GOING SHOPPING

Shopping your sound library to a manufacturer or distributor is worth serious consideration. Many of the companies I spoke with are interested in hearing new ideas for libraries. "A construction kit really works for us," says Doug Rogers of East West Communications. "We like to get one- to four-bar loops of all the instruments, then a breakdown of those loops individually, as well as of the individual instrumental notes."

Korntheuer of Rarefaction says, "I look for collections that are twisted! Something that's not out there already. Something that spurs the imagination and creativity. If we're getting a submission, we like to see most of the programming work already done, such as zero-crossovers, normalization, and so on. We also like to have a multilayered sample broken down into its components, so end users can reassemble it themselves" (see **Fig. 4**).

The process of getting your library heard begins with contacting the company you hope will publish your work. Most of the companies surveyed for this article said that they prefer to get a proposal first before hearing the actual project. If they're interested in your proposal, they will then request that you send material for them to listen to.

Rather than blanketing companies with CD-Rs of your project, do some thorough market research into which companies are supporting your format and what kinds of products they carry. Look at each company's Web page to become familiar with its vision. Also, consider what each company *doesn't* have in its catalog. If your project is an Eastern European bagpipe collection in the Akai format, target companies that emphasize instrumental collections, world music, and Akai CD-ROMs, but that don't already have a release of this sort.

JUST DO IT

The market for sound libraries continues to grow each year, as the demand for sound content escalates in the world of audio and video. The result is an expanding market for unique sound libraries and talented sound designers and programmers. With the right sounds and a good presentation, you can carve out your place among the pros.

Gino Robair *is an Associate Editor at* **EM**.
Thanks to Daniel Fisher, Thomas Dimuzio, Eric Persing, Paul Korntheuer, Jim Miller, Jeff Darby, Melissa Reuther, Doug Rogers, Mark Hiskey, Doug Morton, Jeff Obee, Karen Stackpole, Headless Buddha Labs, Mary Cosola, and Steve Oppenheimer.

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DIGITAL PIPELINES

The introduction of the Alesis ADAT in 1992 was a huge step toward the annihilation of analog technology and the development of the all-digital studio. True, digital audio technology had been around for a while at that point, but it really wasn't until the ADAT and, later, the Tascam DA-88 hit the streets that the majority of musicians began to understand the potential of this technology.

The success of these two machines initiated a quest to improve fidelity in other areas. Soon digital mixers popped up, digital audio workstations gained widespread popularity, digital dynamics processors hit the market, and various digital support gear joined the party.

Today studios on virtually every level can produce an almost entirely digital recording, starting at the sound module/sampler/preamp, going to the console, to the multitrack, through effects processors, and finally to 2-track. Several companies have even started to manufacture microphones and speakers with built-in digital converters, so the digital chain can begin and end right at the transducers. For better or for worse, the all-digital world has arrived.

The mere presence of digital devices, however, does not make a studio completely digital. Sure, the more digital equipment you own, the greater fidelity your recordings will have. But to really capitalize on this technology, you need to have digital connectivity between gear. After all, the signal path is only as strong as its weakest link, and if you're constantly sending audio through converters, from analog to digital and back again, degradation is inescapable.

By Jeff Casey

Everything you ever wanted to know about ADAT Optical and Tascam TDIF.

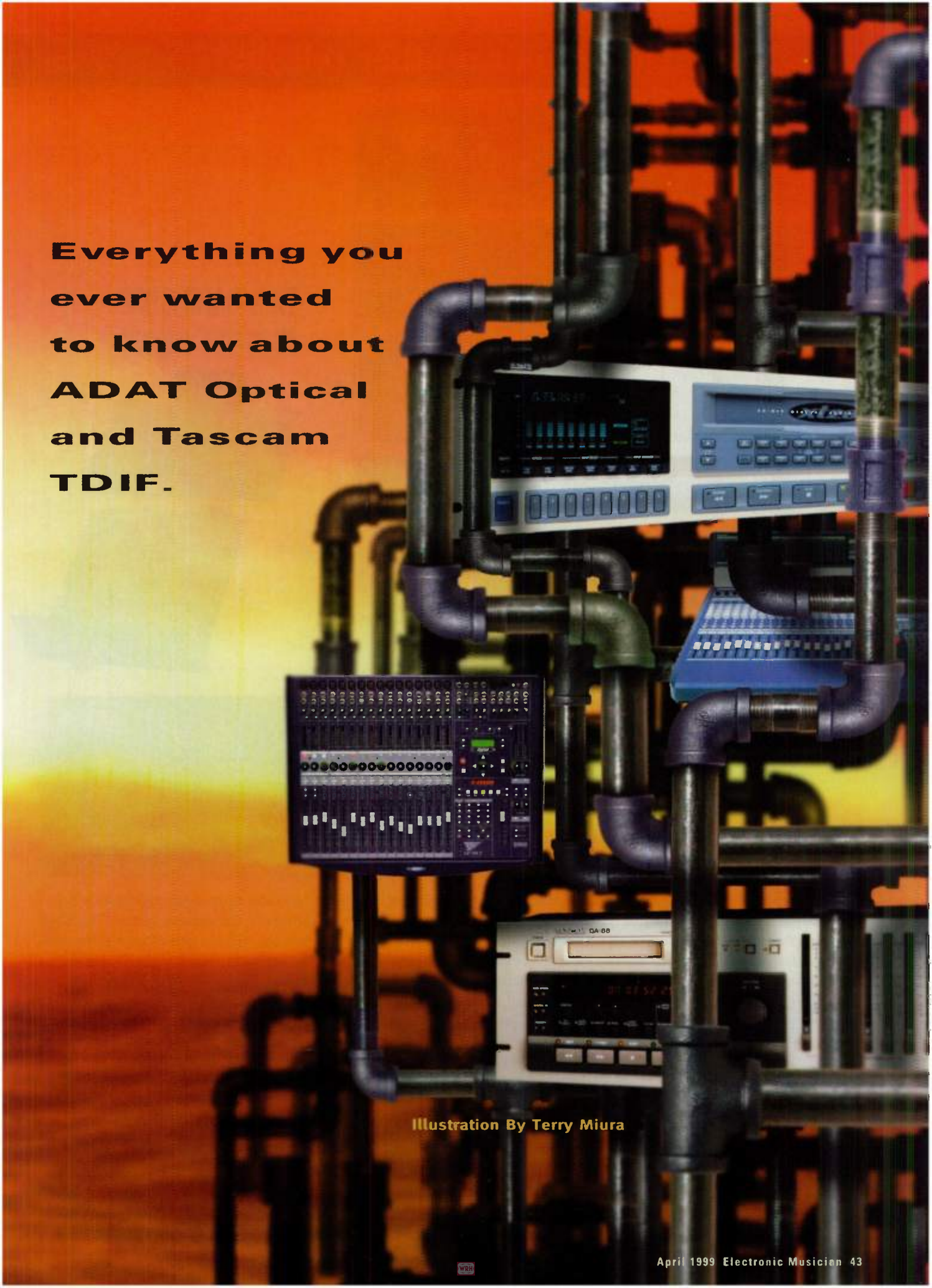

Illustration By Terry Miura

Didn't your mother tell you?

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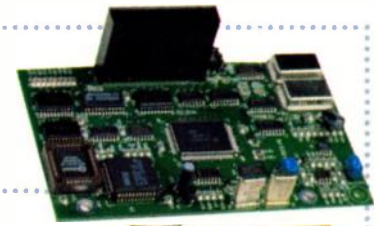

504X

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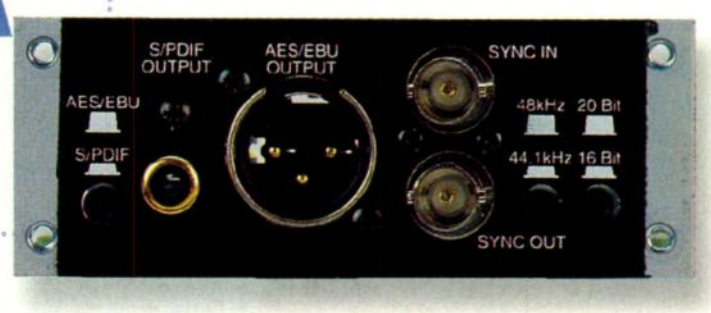
Digital Output System



The 504X represents the pro audio industry's first melding of a vintage 12AU7 vacuum tube design with the clarity and versatility of the finest 24 bit analog to digital conversion process available, the dbx TYPE IV™ Conversion System. The 504X plugs directly into the rear panel of all three dbx Silver Series processors and converts the rich warmth of tube-driven analog signal straight into the digital domain for the first time. It can also be used with the premium 1086 Mic Pre/Processor, for the most economical analog to digital conversion system available today.

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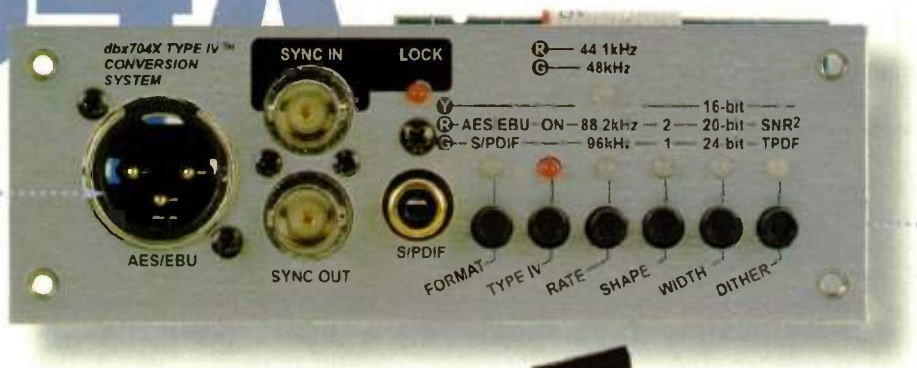
704X/504X

704X

Digital Output System



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Although 2-channel digital transfer formats such as S/PDIF and AES/EBU are a big part of successfully connecting equipment, what really makes the all-digital studio possible is multichannel (that is, more than 2-channel) digital audio transfer formats, specifically ADAT Optical and Tascam's TDIF. These two protocols have revolutionized the industry almost as much as their respective MDMs have. Today people who don't even own ADATs or DA-series recorders are using ADAT Optical and TDIF connections to get from one piece of gear to the next. This month, we take an in-depth look at these two formats, the products that use them, and ways of getting from one format to another.

HISTORY LESSON

Neither ADAT Optical nor TDIF was originally designed to be what it has evolved into. The initial idea behind each was to provide a way to transfer multiple audio tracks simultaneously between identical MDMs (ADAT to ADAT, or DA-88 to DA-88), not between MDMs and other gear.

Our story begins with Alesis circa 1991 and the development of the revolutionary ADAT. While the company was designing the original unit, the good folks at Alesis realized that they needed to provide a means of transferring audio digitally between two machines. After all, the ADAT was to be a modular recorder, intended to connect with others of its kind; it would be silly if users lost a generation every time they made a backup tape or copied tracks to another machine. Alesis knew all too well that critics would have a field day if it didn't come up with a solution.

At the time, AES/EBU and S/PDIF were the only widely used digital audio formats, and they could handle only two channels of audio. Obviously, using either one to transfer eight tracks between ADATs would be cumbersome. So Alesis created its own format that was capable of carrying eight separate signals, and ADAT Optical (also known as Lightpipe) was born. I distinctly remember doing a session in 1993 and being amazed to discover that the ADAT could digitally transfer *all* its tracks from one machine to another at the same time. Wow!

In 1994, Alesis realized that Lightpipe had much potential that was not yet being exploited. The company started offering ADAT Optical connectivity on some of its higher-end synthesizers and signal processors as a way of directly interfacing with an ADAT. Other manufacturers realized that this format could potentially become the de facto standard and wanted to get in on the action. Shortly thereafter, the ADAT Group was formed, consisting of third-party manufacturers working together to build (and capitalize on) the ADAT empire. Today more than 300 companies belong to the ADAT Group, and a good portion of them manufacture products with Lightpipe connectivity.

Over in the competing camp, Tascam wheeled out the DA-88 in 1993, and along with it came the company's answer to Lightpipe, the proprietary Tascam Digital Interface Format (TDIF). That was around the same time that Alesis began contracting with third-party developers. So did Tascam, and a struggle over digital data formats soon ensued—another battle in the war already occurring between the two companies. (For an interesting retrospective look at the state of MDMs in the early '90s, check out "Brave New World" in the October 1993 issue of **EM**.)

Fast-forward five years to 1998. The ADAT and DA-88 have established their places in the annals of audio history, Alesis and Tascam have both expanded their lines of MDMs, and the Lightpipe and TDIF protocols are more popular than ever. Alesis has even created an entire division devoted to developing and promoting ADAT Optical technologies. At this point, Lightpipe connections are certainly more abundant in third-party products than are TDIF connections; that's probably because ADATs are more popular than DA-series recorders in personal studios. In any event, it's almost unheard of for a major digital product to be released today without options for ADAT Optical, and very often TDIF, connectivity.

In a moment, we'll take a look at some of the products that support each format, as well as the many products on the market that change signals from one format to another. But first, let's take a look at the nuts and bolts of the unique format that spawned a revolution.

LIGHTING THE PIPE

Although Lightpipe and TDIF accomplish the same thing, they do so in very different ways. The ADAT Optical protocol works on a 24-bit NRZ (Non-Return-to-Zero) encoding scheme that sends information along a high-bandwidth fiber-optic cable.

What does that mean in English? Basically, each audio channel can be carried at 24-bit resolution (regardless of whether you're actually working with a bit resolution that high). The samples from all eight audio channels are organized into a single block of information called a "data frame," each of which contains 256 bits. If you do the math, you'll discover that this leaves 64 bits that can be used for synchronization and specialized "user" applications (discussed shortly).

This data frame is then sent through a fiber-optic cable by means of a light transmitter (hence the name "Lightpipe")

All Alesis ADATs have 8-channel ADAT Optical I/O, commonly referred to as Lightpipe. The ADAT XT20, shown here, can transfer digital audio with up to 20-bit resolution. With Lightpipe, all eight audio channels are organized into a composite data stream that is sent through a fiber-optic cable via a light transmitter. The receiving unit decodes the signal using a light-detecting element.

and is received in the other unit by a light-detecting element. A single-sided, 5-volt power supply runs the system. When all is said and done, the receiver's output produces an exact replica of the signal that was introduced at the transmitter, save for a nominal propagation delay.

Lightpipe cables typically can run up to about 33 feet, although some people have had success with runs as long as 50 feet. By using more expensive glass cable, you can run ADAT Optical cables quite a bit farther, even without special drivers. Alesis claims that it has successfully tested runs up to 100 feet using these specially terminated glass cables.

One of the greatest benefits of an optical format is the low cost of the cables; in fact, they're so cheap that Alesis gives away a 1-meter cable with every new ADAT. Also, the connections are small enough to fit on a PCI card or digital-mixer card without the need for a breakout box, and the fiber-optic design virtually eliminates the possibility of electromagnetic interference and ground hum.

But the coolest thing about the ADAT Optical format is that after you account for sync bits and various other information bits, four bits are left over in every data frame. Of those, two are reserved for potential future uses to be determined by Alesis (for example, upgrading the system resolution without the need to change formats), and the other two are denoted as "user-definable."

If you need a high-quality, 8-channel A/D converter box with a variety of interfacing options, check out Apogee's AD-8000. The stock unit has 24-bit analog inputs and AES/EBU and S/PDIF I/O. When you add the optional, simultaneously active ADAT Optical and TDIF I/O interfaces, the unit becomes a multipurpose format converter.

This doesn't mean that just any ADAT owners can simply decide what they want to send through the Lightpipe; rather, these bits are available to designers looking to implement Lightpipe in new products. For example, the manufacturer of a remote-controlled digital mic preamp could use the extra bits to send control messages between the stage unit and the mix position.

The ADAT Optical format is free to anyone who wants to incorporate it into a design; all you have to pay for is licensing the ADAT logo, should you decide to use it to market your product. As I mentioned earlier, Alesis recently created a separate division, Alesis Semiconductor, devoted to the development of Lightpipe. Its mandate is to promote ADAT Optical as the industry standard in digital audio transfer. Whether this will actually happen remains to be seen, but it seems like a good bet. Although the company's plans for future Lightpipe implementation are still hush-hush, the live-sound industry appears to be the next frontier. The use of Lightpipe in home-theater systems opens up a great many possibilities, so that direction will probably be explored, too. The future looks exciting for the pro-audio industry and consumers alike.

Keep in mind that although the ADAT Optical specification currently supports 24-bit data, not all gear is capable of handling 24-bit digital audio transfers. Some equipment can send and receive only 20-bit information, a limitation of the Alesis controller chips used to implement Lightpipe in certain products. In fact, Alesis's current crop of ADAT recorders supports 20-bit Lightpipe transfers only.

Akai's DR16 Plus modular hard-disk recorder is essentially a 24-bit version of the old DR16. Stereo AES/EBU and S/PDIF interfaces come standard, and you can add 8-channel Lightpipe I/O, 8-channel TDIF I/O, and 8-channel AES/EBU I/O via optional expansion cards.

T-OTALLY DIF-FERENT

Tascam was reluctant to provide me with information on the specifics of the TDIF format (something about the folks in Tokyo not wanting us to publish their secrets). I searched the Internet, talked to my engineering friends, and begged and pleaded, but I still came up with squat. Tascam keeps this stuff pretty well guarded!

Here's what I do know: whereas the Lightpipe format collapses the multichannel digital information into a single frame of data, TDIF uses a multiwire 25-pin D-sub cable that transmits channel information independently, without creating a composite stream. Obviously, these custom cables are a bit more expensive than optical cable—a 3-foot cable retails for $110, and a 16-foot cable will cost you $135—but they can be run farther than their optical counterparts, albeit with more susceptibility to interference.

THE SUPPORTING CAST

Of course, this article cannot possibly serve as an exhaustive guide to *every* product that supports ADAT Optical and TDIF—there are more than 300 companies manufacturing Lightpipe products alone. But I have compiled a list based on data provided by Alesis and Tascam and from new-product information that I gathered at the 1999 Winter NAMM show in Los Angeles. It's not comprehensive, but I think it's a good representation of the gear that affects most personal-studio owners. The focus here is on digital connectivity—specifically TDIF, Lightpipe, AES/EBU,

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4 Track banks x 10 pads give you 40 tracks of digital audio.

Scratch the ribbon to scratch digital audio just like you would on a turntable. Or, assign the ribbon to control any parameter in realtime.

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Put it all together and you have 42 tracks of digital audio remix automation.

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and S/PDIF options. (Unless noted otherwise, AES/EBU and S/PDIF I/O refer to 2-channel connections.)

MDMs. Ironically, simply interfacing between two identical MDMs doesn't seem very exciting anymore. But being able to connect those machines to the wealth of mixers, DAWs, hard-disk recorders, sound cards, synths, outboard gear, and even competing MDMs that support TDIF and Lightpipe certainly *is* exciting. (Actually, with the advent of onboard digital patch bays, the ability to digitally transfer from one MDM to another is quite cool in itself.)

Lightpipe I/O currently comes standard on the full line of Alesis ADAT recorders, including the ADAT XT20, LX20, and post-production-oriented M-20. ADAT recording is now at 20-bit resolution, but you can set the Type II

PUTTING IT ALL TOGETHER

	Lightpipe Channels	TDIF Channels	S/PDIF Channels	AES/EBU Channels	Analog I/O Channels
Modular Digital Multitrack Tape Recorders (MDMs)					
ADAT LX20	8	n/a	n/a	n/a	8 in/out
ADAT M-20	8	n/a	n/a	8 (optional)	8 in/out
ADAT XT20	8	n/a	n/a	n/a	8 in/out
Studer V-Eight	8	n/a	8 (optional)	n/a	8 in/out
Tascam DA-38	n/a	8	8 (optional)	8 (optional)	8 in/out
Tascam DA-88	n/a	8	8 (optional)	8 (optional)	8 in/out
Tascam DA-98	n/a	8	8 (optional)	8 (optional)	8 in/out
Modular Hard-Disk Recorders (MHDRs)					
Akai DR8	8 (optional)	8 (optional)	2	2; 8 (optional)	8 in/out
Akai DR16 Plus	16 (optional)	16 (optional)	2	2; 8 (optional)	16 in/out
E-mu Darwin	8 (optional)	n/a	2	n/a	8 in/out
Fostex D-90	8	n/a	2	n/a	8 in/out
Fostex D-160	16	n/a	2	n/a	16 in/out
Digital Mixers					
Mackie D8B	up to 24 (optional)	up to 24 (optional)	2	2	various
Panasonic WR DA7	up to 24 (optional)	up to 24 (optional)	2; up to 24 (optional)	4; up to 24 (optional)	various
Spirit Digital 328	16 in; 24 out	16	2	2	various
Tascam TM-D1000	8; 8 (optional)	n/a	4 (optional)	2 outs; 4 (optional)	various
Tascam TM-D4000	up to 24 (optional)	up to 24 (optional)	2	2; up to 24 (optional)	various
Tascam TM-D8000	n/a	24 tape returns; up to 24 inputs	up to 24 inputs	up to 24 inputs	various
Yamaha 01V	8 (optional)	8 (optional)	2	8 (optional)	various
Yamaha 02R	up to 32 (optional)	up to 32 (optional)	2	2; up to 32 (optional)	various
Yamaha 03D	8 (optional)	8 (optional)	2	2; 8 (optional)	various
Audio Cards/DAWs					
Alesis ADAT/PCR	8	n/a	n/a	n/a	n/a
Digital Audio Labs V8	8 (optional)	8 (optional)	n/a	2 (optional)	8 (optional)
Digidesign Pro Tools	up to 64 (optional)	n/a	n/a	up to 64 (optional)	up to 64 in/out (optional)
Ensoniq PARIS with MEC	up to 72 (optional)	n/a	2	n/a	4 in/out; up to 74 in or 74 out (optional)
Korg 1212 I/O	n/a	8	2	n/a	2 in/out
Lexicon Studio with LDI-16S	8 (optional)	8	n/a	8 (optional)	8 in/out
Merging Technologies Pyramix Virtual Studio with Keops card	16	16 (optional)	2; 4 switchable w/ Lightpipe	4 (optional)	4 in/out (optional)
MOTU 2408	24	24	4	n/a	8 in/out
Mytek DAW 9624	16	8 (optional)	n/a	8	8 in/out
Otari RADAR II	n/a	24	n/a	n/a	24 in/out
SEK'D Prodif Gold	8	n/a	2	n/a	n/a
Sonorus StudI/O	16	n/a	2	n/a	n/a
Soundscape Mixtreme	n/a	16	2	n/a	n/a
Soundscape SSHDR-1 with SS8IO-1	8	8	2	n/a	8 in/out
Synthesizers					
Alesis QS7	n/a	4 out	n/a	n/a	4 out
Alesis QS8	n/a	4 out	n/a	n/a	4 out
Korg Trinity series with TR-Rack	n/a	4 out (optional)	n/a	n/a	4 out
Kurzweil K2500 with DMTi	8 (optional)	8 (optional)	2 (switchable w/ AES/EBU pair)	4 in/8 out (optional)	8 out (optional)
Other Gear					
Alesis Q2	2	n/a	n/a	n/a	2
Aphex 1788	8 out (optional)	8 out (optional)	n/a	8 out (optional)	8 out
Digital Labs Fiber Optic Patch Panel	8	n/a	n/a	n/a	n/a
Drawmer 1962	2 out	n/a	n/a	n/a	n/a
Studer D19 MicA/D	2 out	n/a	n/a	n/a	n/a
Studer D19 MicValve	2 out	n/a	n/a	n/a	2 out

For manufacturer information, please see Contact Sheet, p. 163.

The biggest thing in samplers wouldn't even fit on this page

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- Up to 256 Mb RAM
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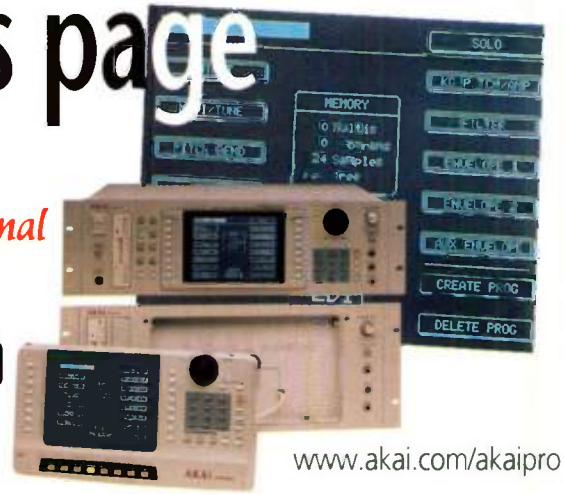

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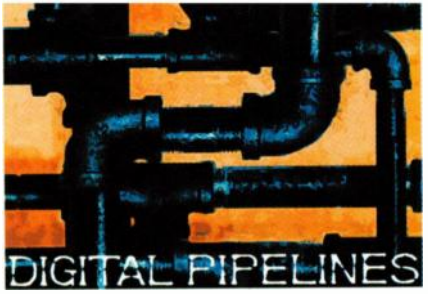


recorders to produce a 16-bit output signal complete with dither so that they're compatible with the older ADAT-XT and "blackface" models.

Other MDM manufacturers—not surprisingly, companies that have developed S-VHS recorders based on the ADAT—also offer Lightpipe connectivity. For example, the 20-bit Studer V-Eight offers eight channels of ADAT Optical I/O. If you feel like dusting off a 16-bit Fostex CX8 or RD8, each of those units provides standard Lightpipe ports, as does the Panasonic MDA-1. (The Fostex and Panasonic MDMs are no longer manufactured, so they're not included in the table "Putting It All Together.")

On the Tascam front, 8-channel TDIF I/O is standard fare on all of the DA-series multitracks, including the DA-38, the DA-88, and the ultrahip DA-98. All three Tascam decks can interface with the AES/EBU format via the optional IF-88AE.

Interestingly, you'll find no TDIF connections on the discontinued Sony version of the DA-88, the 16-bit PCM-800; there are only eight channels of AES/EBU. Not surprisingly, Alesis doesn't offer a TDIF option for ADATs, and Tascam doesn't provide Lightpipe connectivity. Tascam, however, recently released a nifty TDIF/Lightpipe converter, the IF-TAD.

MHDRs. The ability to digitally connect a modular hard-disk recorder (MHDR) with an MDM can be a valuable asset to the personal-studio owner. For example, you might want to use the MDM as your main recording medium, fly tracks out to the MHDR for editing, and then fly them back to the MDM, all the while maintaining sync between the decks. Or you could do all your work within the MHDR and use the MDM just for backup purposes. Many people use a combination of MDM and MHDR simply to double their track count.

Akai, Fostex, and E-mu are leading manufacturers of MHDRs, and all three companies offer Lightpipe I/O options for their units. The Fostex D-90 offers eight channels of Lightpipe I/O as a standard feature, and the D-160 has two Lightpipe ports for interfacing with a 16-track ADAT system. Both 16-bit units offer optical S/PDIF I/O for backing up data to DAT. E-mu's 16-bit Darwin recorder has an 8-channel Lightpipe option in addition to the S/PDIF I/O that ships with the standard unit.

Akai's machines offer both ADAT Optical and TDIF connections. The Akai DR8 and DR16 Plus recorders have options for 8-channel Lightpipe I/O, 8-channel TDIF I/O, and 8-channel AES/EBU I/O. Connections for AES/EBU and S/PDIF come standard with both decks.

Roland's DM-800, predecessor to the company's popular VS line of production systems, had an option that allowed 8-channel TDIF and Lightpipe connections. So, if you can find a DM-800, it may or may not be outfitted for multitrack digital audio transfer. For those folks who have $24,950 to spend, Otari's 24-track RADAR II system offers 24 channels of TDIF I/O.

Digital mixers. In any studio, audio is constantly being routed between the multitrack and the mixing console, so maintaining digital integrity is probably more important here than anywhere else in the signal path. That's why most digital mixers offer an abundance of connectivity options.

The Panasonic WR-DA7's digital I/O must be configured from scratch by adding expansion cards. It has three card slots, each of which can accommodate an 8-channel ADAT Optical, TDIF, or AES/EBU (hardware switchable to S/PDIF) card. There is also an 8-channel A/D/A card, and the mixer provides switchable AES/EBU and S/PDIF 2-track inputs and record outputs (on XLR jacks). Furthermore, aux sends 1 and 2 employ S/PDIF on RCA jacks.

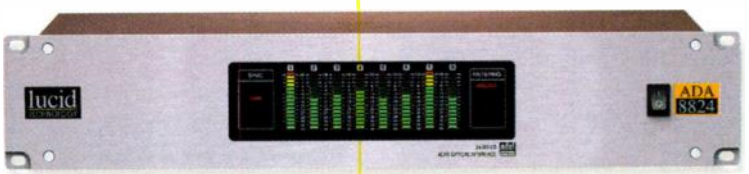
Sporting just as many I/O options, the Yamaha 02R offers four expansion slots that accept 8-channel Lightpipe, TDIF, and AES/EBU cards. The stock mixer also comes with AES/EBU and S/PDIF I/O. The 02R's smaller cousin, the Yamaha 03D, offers the same card options but has only one available slot, as does the even smaller 01V. The 03D provides both AES/EBU and S/PDIF I/O, while the 01V offers S/PDIF only. The long-anticipated Mackie Digital 8•Bus (D8B) also has three card slots for adding Lightpipe or TDIF inputs, and it comes standard with AES/EBU and S/PDIF connections.

The Tascam TM-D1000 comes standard, of course, with 8-channel TDIF I/O. The TM-D1000 also gives you two XLR AES/EBU outputs; an optional expansion card adds eight more channels of TDIF plus four channels of AES/EBU or S/PDIF. The Tascam TM-D4000 ships with S/PDIF and AES/EBU I/O and allows you to install up to three 8-channel cards for additional TDIF, ADAT Optical, AES/EBU, or analog I/O. Tascam's top-of-the-line console, the TM-D8000, provides 24 dedicated TDIF tape returns and inputs that can be configured to accept analog, TDIF, S/PDIF, or AES/EBU signals.

The Spirit Digital 328 offers 16 channels of ADAT Optical input, 24 channels of Lightpipe output, and 16 channels of TDIF I/O on the base unit, along with AES/EBU and S/PDIF I/O. Similarly, the now-discontinued Korg 168 RC provides 16 channels of Lightpipe I/O and stereo S/PDIF, but it has no TDIF or AES/EBU ports. The 24-bit Roland VM-3100 will have an optional interface box that adds both Lightpipe and TDIF connections.

Finally, for those with fat wallets, RSP Technologies' Project X ($3,500 for a basic system) can be outfitted with both Lightpipe and TDIF cards, as well with an 8-channel AES/EBU card or several A/D/A cards.

Audio cards/DAWs. Although you'd think that a digital audio workstation is



Lucid Technology's ADA8824 is a multiformat, 24-bit converter that offers eight channels of analog I/O, eight channels of ADAT Optical I/O, eight channels of AES/EBU I/O, and stereo S/PDIF I/O.

Secrets of the DAW Masters

"Some of the most distinctive processors anywhere."
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Plug-Ins For Audio Production

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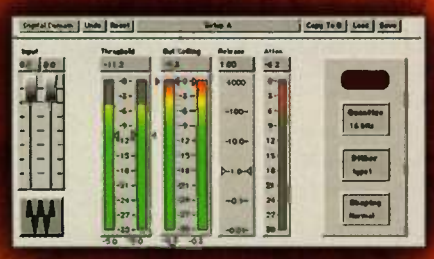
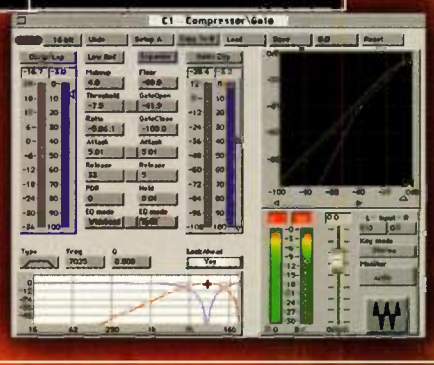


C1 UNPARALLELED PARAMETRIC COMPANDER



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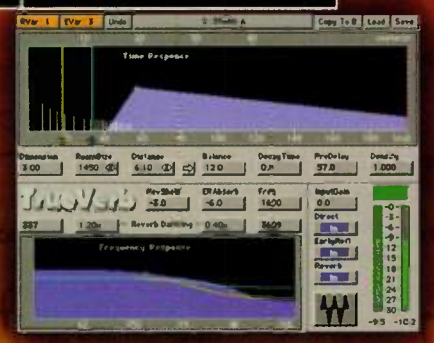
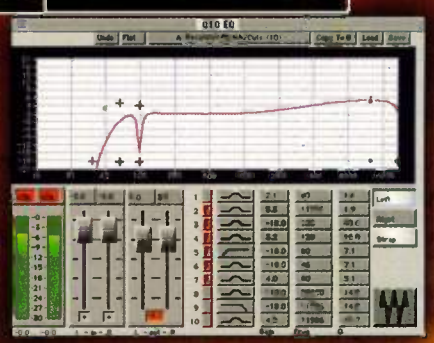


Q10 SURGICAL EQUALISATION



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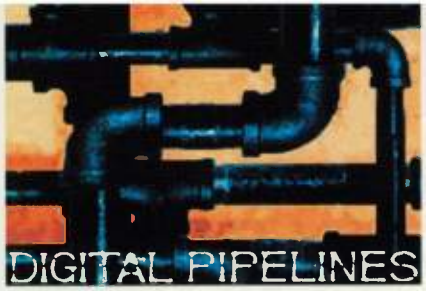

a self-contained environment, in reality it rarely is. Many people prefer to use DAWs like MHDRs, employing them as editing and processing tools rather than as comprehensive recording systems. Others like to back up data to a particular MDM format simply because random-access storage can get to be a bit expensive. And then there's the fact that any facility set up to work on outside projects should be equipped to handle whatever tape format the client might provide. So even if you use a DAW, chances are you'll need to connect to other gear.

For the past five years, Digidesign's Pro Tools system has remained the most popular Macintosh-based DAW for audio production. Pro Tools is a customizable working environment, and to get any I/O with it, you have to buy interface modules. Pro Tools can interface directly with an ADAT system by using the Digidesign ADAT I/O Bridge, a rack-mount unit that provides 16 channels of Lightpipe I/O. Digidesign does not currently make a dedicated TDIF module, but that doesn't mean you can't use Pro Tools with, say, a DA-88. You'd just need to have a format converter to translate the TDIF signals to optical or AES/EBU and then fly the audio into the appropriate Digidesign interface (the ADAT I/O Bridge for Lightpipe, or the 888 for AES/EBU). The same, of course, is true for any piece of gear discussed here that doesn't support a particular format.

A strong entry into the cross-platform DAW market last year was Ensoniq's PARIS. Like Pro Tools, PARIS offers a customizable I/O system in the form of a modular expansion chassis. The Interface MEC can be loaded with up to nine 8-channel A/D, D/A, or ADAT Optical cards. S/PDIF I/O is standard on the MEC, as well as on most other PARIS interfaces. The A/D and D/A cards both offer 24-bit conversion.

The cross-platform Lexicon Studio accommodates a number of digital formats. As of this writing, two I/O modules are available: the LDI-12T offers two channels of analog I/O and two channels of switchable S/PDIF/Lightpipe I/O, while the LDI-16S provides eight channels of both TDIF and analog I/O. Options are available for the LDI-16S that add eight channels of AES/EBU I/O or eight channels of ADAT Optical I/O.

The PC-based V8 workstation from Digital Audio Labs can use optional PCI I/O cards for interfacing with both Lightpipe and TDIF devices. Multiple I/O cards may be cascaded, and the

CONVERTERS COMPARED

	Lightpipe Channels	TDIF Channels	S/PDIF Channels	AES/EBU Channels	Analog I/O Channels
Alesis AI-3		n/a	n/a	n/a	8 (20-bit A/D/A)
Apogee AD-8000	8 (optional)	8 (optional)	8 channels selectable between S/PDIF and AES/EBU		8 (24-bit A/D/A)
Apogee FC-8	8	8	n/a	n/a	n/a
Apogee Rosetta	2	2	2	2	2 (24-bit A/D/A)
Graham-Patten ADAT-1/ADAT-2	8	n/a	n/a	n/a	8 (20-bit A/D/A)
Graham-Patten ADAT-3/ADAT-4	8	n/a	n/a	8	n/a
Korg 880 A/D	8	n/a	n/a	n/a	8 (18-bit A/D)
Korg 880 D/A	8	n/a	n/a	n/a	8 (18-bit D/A)
Kurzweil DMTi	8 (optional)	8 (optional)	2 (switchable with an AES/EBU pair)	4 in/8 out	n/a
Lucid Technology ADA8824	8	n/a	2	8	8 (24-bit A/D/A)
Midiman Pipeline 8x8	8	n/a	n/a	n/a	8 (24-bit A/D/A)
Midiman SAM	8	n/a	2	n/a	n/a
MusicNet AD24 and DA24	8	n/a	n/a	n/a	8 (24-bit A/D/A)
Otari UFC-24/8	24	24	n/a	8 (upgradable to 24)	n/a
Prism Sound MR-2024T	n/a	8	2 (assignable)	8	n/a
Sonorus AudI/O AES/8	8	8	8 channels selectable between S/PDIF and AES/EBU		n/a
Sonorus AudI/O AD/9624 and DA/9624	8	8 (optional)	n/a	8	8 (24-bit A/D/A)
Sonorus AudI/O AD/24 and DA/24	8	n/a	n/a	n/a	8 (24-bit A/D/A)
Soundscape SS8IO-1	8	8	n/a	n/a	8 (20-bit A/D/A)
Soundscape SS8IO-2	8	8	n/a	n/a	n/a
Soundscape SS8IO-3	n/a	8	n/a	n/a	8 (20-bit A/D/A)
Spectral Translator	8	8	n/a	8 (Translator Plus)	n/a
Studer D19 MultiDAC	8 (optional)	8 (optional)	n/a	8	8 (24-bit D/A)
Tascam IF-88AE	n/a	8	n/a	8	n/a
Tascam IF-TAD	8	8	n/a	n/a	n/a

For manufacturer information, please see Contact Sheet, p. 163.

Finally, A Sequel That's Actually Better Than The Original

Alesis synthesizers

The QS6.1's four real-time control sliders are assignable to any mod destination, including envelopes, LFOs and even multieffects.

With two expansion ports, the QS6.1 can access another 16MB of sounds for a total of 32 meg available at once. Use our QCard™ expansions in your musical style of choice, or burn your own samples to a Flash RAM card using the included Sound Bridge™ software.

It doesn't usually happen this way. Sequels are supposed to be boring and derivative. But the new QS6.1™ takes the powerful 64 voice synth engine of the original QS6 and supercharges it with double the sound memory, double the expansion capacity, new performance features and much more. So how is it that the QS6.1 got a whole lot better than the keyboard it replaced while actually costing less? The answer is that this sequel is from Alesis – the company that always delivers more than you expect.

QS6.1™

64 VOICE EXPANDABLE SYNTHESIZER

QS6.1 New Features

- Double the sound ROM of the QS6 (16MB internal)
- Now includes Alesis' stereo grand piano sounds from the QS8
- Enhanced GM sound set
- Double the expansion capacity (up to 32MB total)
- Four control sliders
- Big new LCD display
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ALESIS

V8 provides full transport control of ADAT and DA-series recorders directly from software. AES/EBU and S/PDIF I/O options are available as well.

Soundscape's integrated PC-based SSHDR-1 hard-disk recording system comes standard with eight channels of TDIF I/O and two channels of S/PDIF I/O. But to send and receive Lightpipe signals, you must have an additional I/O module. The company's Mixtreme PCI audio card provides 16 channels of TDIF; however, you'll still need an external breakout box to gain access to Lightpipe and other types of I/O. There are three I/O modules that can be connected to either system: the SS8IO-1 (balanced analog/Lightpipe/TDIF), the SS8IO-2 (TDIF/Lightpipe), and the SS8IO-3 (unbalanced analog/TDIF), all of which will be discussed later. TDIF ports are used to connect all Soundscape hardware.

The Otari PD-80 digital audio workstation has options for TDIF and Lightpipe and comes standard with AES/EBU I/O. Otari's RADAR II system provides TDIF connections on the base unit, and you can gain ADAT Optical and AES/EBU I/O with the assistance of the company's stand-alone UFC-24 format converter. The Pyramix Virtual Studio from Merging Technologies also offers TDIF and Lightpipe I/O options.

As part of its Soundlink line of products (which includes the 168 RC mixing console), Korg has introduced the 1212 I/O, a cross-platform PCI audio card that offers 8-channel ADAT Optical I/O, as well as 2-channel analog and S/PDIF I/O. More analog I/O channels can be added using the company's 880 A/D and 880 D/A, both of which can also function as stand-alone Lightpipe-to-analog converters.

Along the same lines, Sonorus offers the StudI/O PCI audio card, which provides two sets of optical I/O parts that can be independently switched between stereo S/PDIF and 8-channel ADAT Lightpipe formats. When used in combination with one of the company's hardware AudI/O units, the StudI/O can accept AES/EBU, TDIF, and analog signals. SEK'D recently rolled out its Prodif Gold ADAT card, which brings Lightpipe and S/PDIF signals into and out of a PC.

Also taking advantage of the Lightpipe format is Mytek, best known for its high-end mastering converters. Through a joint venture with Sonorus, it has developed the expandable DAW 9624, which consists of a StudI/O card and the Mytek 8X96-series 24-bit/96 kHz 8-channel A/D and D/A converters. In addition, the Mytek units provide conversion to and from AES/EBU and TDIF (optional). The DAW 9624 is yet another great way to bring analog signals into an ADAT system.

One of my favorite devices is the MOTU 2408. This ingenious box can serve either as the front end to a DAW or as a stand-alone audio-format converter. The 2408 supports 24 channels of I/O that can use any combination of its three sets of TDIF ports, three sets of Lightpipe ports, two sets of S/PDIF ports, and eight analog inputs and outputs. A PCI card allows direct connection to either a Mac or a PC for use with a variety of software applications, and you can connect additional 2408s to your system for a total of 72 channels of simultaneous I/O.

Lastly, the cross-platform ADAT Edit package from Alesis simply provides eight channels of Lightpipe I/O on an ADAT/PCR PCI card. It does not come with other types of digital or analog I/O.

Synthesizers. It seems only logical that synths with multiple outputs should take advantage of TDIF and Lightpipe. Well, it's taken a while, but manufacturers are now coming around. Obviously, Alesis would be foolish not to implement Lightpipe on its synths: in the company's current product line, the QS7 and QS8 have ADAT Optical outputs.

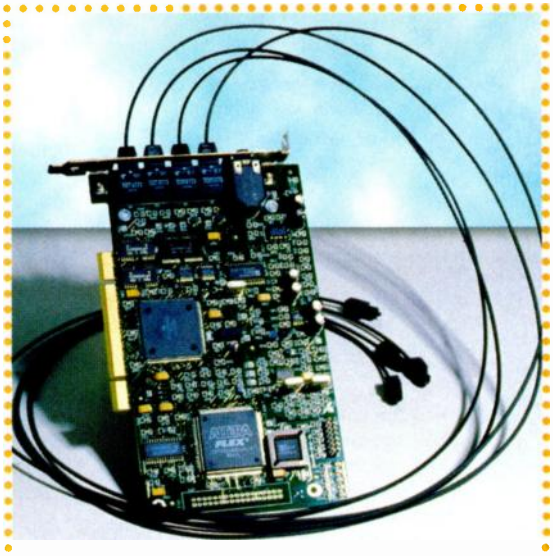
Korg's Trinity line of workstations, including the Trinity, Trinity Pro, and Trinity ProX, offer Lightpipe output options. And using the Kurzweil DMTi format converter, you can interface a K2500 with both Lightpipe and TDIF devices. I think we'll see a lot more of such products in the next few years.

Other gear. Although TDIF and Lightpipe are designed to carry multiple audio channels, they can also serve lesser functions. Several companies have recently started incorporating these formats into their multi-effects processors and preamps.

Currently, you can find ADAT Optical and TDIF outputs on several mic preamps, including the Drawmer 1962 tube preamp, which delivers a 24-bit signal in either format at the output. The Aphex 1788 8-channel mic preamp can also be outfitted to connect with TDIF- and Lightpipe-friendly units (a great solution for live recording).

The Studer D19 MicValve 2-channel mic/line preamp card for the modular D19 series offers ADAT Optical output and 20-bit D/A conversion. The company's D19 MicA/D for the same series also supports Lightpipe. Alesis's Q2—a descendant of the Quadraverb, the first non-ADAT machine to employ Lightpipe—provides ADAT Optical I/O, as well.

Digital Labs has pioneered quite a neat device: the first Fiber Optic Patch Panel for ADAT Optical and S/PDIF optical formats. This box allows you to patch digital signals just as you would on an analog patch bay. The single-rackspace panel has eight fiber inputs and outputs on both the front and rear panels. Also included is a



The Sonorus StudI/O PCI card has two independent sets of optical inputs and outputs. ADAT Lightpipe and stereo optical S/PDIF use the same type of connector, and Sonorus lets you independently configure each I/O pair to handle either type of digital transfer.

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CONFOUNDING CONVERSION

It's great to be able to stay in the digital domain throughout the recording and mixing process, but what happens when you need to send eight or more channels from one multitrack format to another, or to analog, or to a completely different digital format such as AES/EBU or S/PDIF? Several companies make format converters, and they come in all shapes and sizes. What follows here is the lowdown on the current crop of useful little boxes that will get you from one format to another.

But remember, it's not always necessary to have a stand-alone converter to go from format to format. Other devices (especially mixers and DAWs) can often be used to convert signals, provided they offer compatibility with the formats you're working with.

For example, if you have a Panasonic WR-DA7 with a sufficient number of expansion cards, you can generally get wherever you need to go, be it from Lightpipe to TDIF, Lightpipe to S/PDIF, TDIF to AES/EBU, or TDIF to analog. The same is true of the Yamaha series of digital recording consoles: an 02R or 03D makes a great centerpiece for a studio, simply because of all the I/O options it provides. (For a comparative look at the gear discussed in this article, including the digital formats supported, refer to the tables "Putting It All Together" and "Converters Compared.")

Alesis AI-3. The Alesis AI-3 ($499) is a stand-alone 8-channel A/D/A converter that translates signals to and from the Lightpipe format. The AI-3 provides eight analog inputs and outputs on balanced ¼-inch TRS connectors. The converters are 20-bit with 128-times oversampling and deliver a dynamic range of 96 dB. The front panel comes with signal-present and peak LEDs for each channel, and the unit can be switched between +4 dBu and -10 dBV operation.

Apogee AD-8000. The AD-8000 ($5,995) is a comprehensive 8-channel conversion solution, capable of interfacing with ADAT Optical, TDIF, AES/EBU, S/PDIF, and the analog world. It features eight 24-bit A/D line-level converters that boast a dynamic range of 114 dB.

The AD-8000 outputs the converted signals through Lightpipe (optional), TDIF (optional), or eight channels of either AES/EBU or S/PDIF (selectable). The unit has stereo S/PDIF and AES/EBU inputs as well, and options provide an additional two or eight analog outputs. The AD-8000 can also be used as a multiple-format conversion system to transfer signals between all available interfaces simultaneously. And because all outputs are active at once, the AD-8000 makes a great distribution system.

The AD-8000 features the patented UV22 process for translating high-resolution digital audio to 16 or 20 bits without losing quality. In addition to the standard 44.1 and 48 kHz sampling rates, the AD-8000 locks intelligently to word clock or any digital input, including video (with the optional video synchronization card).

Apogee FC-8. The FC-8 ($549) is a reasonably priced transfer system for moving between TDIF and ADAT Optical. The unit is bidirectional, so the TDIF and ADAT machines can each act as either master or slave, and the FC-8 will simultaneously transfer signals in both directions. Housed in a freestanding enclosure about the size of a DI box, the FC-8 has three LEDs that indicate Lock, TDIF Active, and ADAT Active. In addition to the 25-pin TDIF port and the optical port, there's a BNC connector that provides word-clock output.

Apogee Rosetta AD. The Rosetta AD ($1,295) is a 2-channel, 24-bit A/D converter that outputs either ADAT Optical, TDIF, AES/EBU, or S/PDIF—all are standard on the base unit. (Apogee is planning a companion unit, the Rosetta DA.) There are also two AES/EBU outputs, so two signals can always be fed simultaneously from the unit. The Rosetta AD features balanced XLR connections and has an impressive dynamic range of 120 dB. Sampling rate on the standard unit is switchable between 44.1 and 48 kHz, but as an option you can also get the Rosetta AD with selectable 88.2 and 96 kHz rates ($1,995). An A/D/A version, the PSX-100 ($2,995), is also available.

Graham-Patten ADAT Interfaces. This line of converters is designed for use with computer sound cards that offer only ADAT Optical connections. Four models are available.

The ADAT-1 ($1,499) converts eight channels of analog audio into a Lightpipe signal. Input connectors are ¼-inch TRS jacks, and level trims are provided for each of the inputs. A/D conversion is 20-bit. Audio sampling and the ADAT output can be internally referenced or synchronized to word clock or to an AES3-ID signal from the reference input. Two versions are available: one that operates at 44.1 kHz and another that works at 48 kHz.

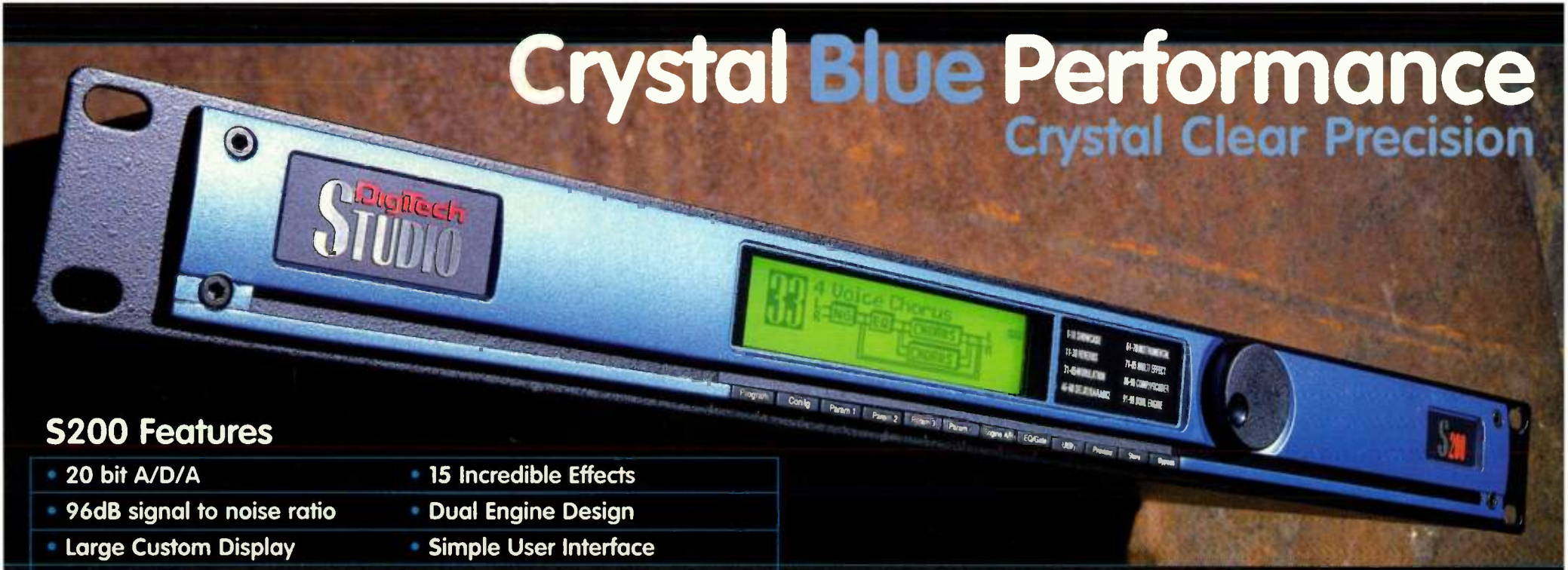
The ADAT-2 ($1,499) is the D/A companion to the ADAT-1. It adds a stereo headphone output with associated volume control to allow monitoring of any channel pair. There is also a sync output providing either a silent AES3-ID signal or word-clock reference.

The ADAT-3 ($899 to $1,299) takes four AES/EBU pairs and converts them

Tascam's DA-series MDMs (including the DA-88 shown here) use the company's proprietary TDIF digital audio interface, which transmits channel information independently, without creating a composite stream. Unlike the competing Lightpipe format, TDIF does not use inexpensive optical cables; instead, it uses a costlier multiwire cable with a 25-pin DB25 connector.

Crystal Blue Performance

Crystal Clear Precision



S200 Features

- 20 bit A/D/A
- 96dB signal to noise ratio
- Large Custom Display
- True Stereo Operation
- 15 Incredible Effects
- Dual Engine Design
- Simple User Interface
- 99 User and 99 Factory Presets

Each menu in the S200 features simple graphics for effortless operation. The easy-to-use interface doesn't overwhelm you with confusing screens of endless parameters. The S200 lets you get right to the business of making music, not wasting time programming.

Behind the cool blue exterior of the Digitech S200 lies the impressive clarity of an honest 20 bit A/D D/A conversion, coupled with an ultra-quiet -96dB signal to noise ratio, allowing you to run discreet, crystal clear signal paths. The powerful dual engine processor of the S200 enables you to use any one of the five different effect configurations capable of placing effects in any order. The large custom LCD interface makes any effect or parameter easily accessible. Whether you are in the studio or on the stage, The Digitech S200 has just the right color for any situation.

Other Studio Series 20 bit Processors Include:

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into a single 8-channel Lightpipe stream. It has a reference input/output connector and a 9-pin D-sub miniature remote-control connector. The ADAT output can be synched to any one of the main AES inputs or to word clock or a secondary AES3-ID sync signal fed to the reference input. With the ADAT-3 synched to one of the AES inputs, you can also use the reference connector to output word clock. The control connector allows you to individually mute each input pair and identifies which of the main AES inputs is being used as the sync source. Six models of the ADAT-3 are available. The ADAT-3/A0, ADAT-3/A2, and ADAT-3/A4 have AES/EBU inputs on XLR connectors, with zero, two, or four sample-rate converters, respectively. The ADAT-3/B0, ADAT-3/B2, and ADAT-3/B4 have AES3-ID sync inputs on BNC connectors, with zero, two, or four sample-rate converters, respectively.

The ADAT-4 ($899) performs the reverse function of the ADAT-3. Here, Lightpipe signals are converted back into four AES/EBU pairs. The AES outputs are synchronous with the ADAT input, and there is also a sync connector providing a silent AES3-ID sync or word-clock output. Two versions are available: the ADAT-4/A uses XLR connectors, while the ADAT-4/B employs BNC connectors.

Korg 880 A/D and D/A. The Korg 880 A/D ($950) and 880 D/A ($800) are designed for use with either the 168 RC or 1212 I/O to expand the analog I/O of those units via ADAT Optical. However, you can also use these gizmos with any device that supports Lightpipe. Both units offer eight balanced analog connections on ¼-inch TRS jacks, with 18-bit A/D and 20-bit D/A converters.

Kurzweil DMTi. The DMTi ($1,310) is another converter designed for use with a specific product—in this case with the Kurzweil K2500. But it works well as a stand-alone converter for translating between ADAT Optical, TDIF, AES/EBU, and S/PDIF.

On the input side are eight channels of K2500 protocol, eight channels of TDIF or Lightpipe (optional), and two stereo channels of AES/EBU (one of which is switchable to a stereo channel of S/PDIF optical). Outputs include four stereo AES/EBU channels (or three stereo AES/EBU and one stereo S/PDIF optical), eight ADAT or TDIF channels (optional), and eight Kurzweil Digital Stream channels. In order to transfer directly between TDIF and Lightpipe, two DMTi's are required. Any of the outputs can be synched to an external clock or to 44.1 or 48 kHz internal rates.

Lucid Technology ADA8824. The latest offering from this high-flying converter company is the ADA8824 ($3,295), a 24-bit A/D and D/A interface designed for Lightpipe systems. The ADA8824 changes eight analog signals into ADAT Optical (and vice versa) using delta-sigma converters that yield a dynamic range of 113 dB; special digitally controlled attenuators on the inputs and outputs help the user achieve the greatest dynamic range possible. Connections are made via balanced XLR jacks. There are also eight channels of AES/EBU I/O and an S/PDIF I/O. The 2-rackspace unit features discrete LED meters for each channel.

Midiman Pipeline 8×8. The Pipeline 8×8 ($899.95) is an 8-channel, 24-bit conversion system that takes analog signals and outputs them in Lightpipe format (or vice versa). The Pipeline provides balanced and unbalanced analog I/O, selectable word widths, MIDI I/O, word-clock I/O, and selectable internal sample rates of 44.1 and 48 kHz. The Pipeline also features patented Bit-Razor technology, which automatically encodes and decodes 24-bit audio on two ADAT tracks, turning any ADAT into a 24-bit recorder.

Midiman SAM. The other offering from Midiman is the SAM ($399.95), the "S/PDIF ADAT mixer." It accepts Lightpipe and S/PDIF I/O, providing conversion between the two formats. But with the SAM, you can also mix all eight ADAT Optical signals down to a stereo pair of S/PDIF, using the individual channel gain and pan controls included on the unit.

MusicNet AD24 and DA24. Available from Wave Distribution, the half-rackspace MusicNet AD24 ($749) and DA24 ($549) provide 24-bit analog-to-Lightpipe and Lightpipe-to-analog conversion, respectively. All analog connections for the two units are via XLR jacks, and both support 44.1 and 48 kHz sampling rates. The DA24 also is switchable between 16- and 24-bit conversion.

The AD24 can generate or sync to word clock, while the DA24 only generates word clock. The AD24 features a Calibration mode that keeps the converter's internal modulators and integrators properly matched. The DA24 includes a deemphasis filter that eliminates the high-frequency boost that is often used with commercial CDs.

Otari UFC-24/8. The UFC-24/8 ($2,595) universal format converter from Otari can transfer up to 24 channels at a time to and from PD, SDIF-II, Lightpipe, TDIF, and AES/EBU formats. (There are 8 AES/EBU channels on the base unit, and it can be retrofitted for 24 channels). You can link multiple units to provide sample-accurate transfer of more than 48 channels.

All outputs (except what's being used as the source) are active simultaneously, so the UFC-24/8 also makes a terrific distribution system. MIDI I/O is provided for control of routing presets and input-format selection. You can save all parameters of the UFC-24/8, as System Exclusive data, to a MIDI storage device and later recall it.

Prism Sound MR-2024T. The Prism MR-2024T ($2,570) is designed to bring AES/EBU connectivity to Tascam DA-38, DA-88, and DA-98 recorders. The MR-2024T converts between TDIF and AES/EBU or S/PDIF (selectable) and offers some other features aimed at high-quality applications.

The recording mode can be set to eight 16-bit tracks, six 20-bit tracks, four 24-bit tracks, or two tracks at 24-bit and 96 kHz. The MR-2024T tags the tape so that the correct mode is always selected on playback. You can connect a second Tascam machine to the same MR-2024T as a security backup; a monitor selector is provided to determine the playback machine.

Soundscape's SSHDR-1 hard-disk recorder comes with 8-channel TDIF I/O and stereo S/PDIF. Expansion modules include the SS8IO-2 (shown here), which adds Lightpipe I/O, a second TDIF port, and word clock and Superclock connections.

For manufacturer information, please see Contact Sheet, p. 163.

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KEYFAX Hardware

The MR-2024T's synchronization options are versatile. Using DA-88 sync, the transport can run free or locked to video or word clock while the unit provides sync outputs. Alternatively, the MR-2024T can synchronize the DA-88 to an AES3 or S/PDIF input. You can slave up to 16 machines, for a total of 96 tracks at 20-bit or 64 tracks at 24-bit.

Sonorus AudI/O. Sonorus recently introduced a line of converters intended to work as both stand-alone units and interfaces for the company's StudI/O digital audio card.

First up is the AudI/O AES/8 ($999), which has eight channels of TDIF, Lightpipe (used to network with the StudI/O card), and AES/EBU (switchable to S/PDIF) I/O. The outputs are not simultaneously available, so the AES/8 can't operate as a distribution device.

Also available are the AD/9624 and DA/9624 ($2,995 each), both high-quality 24-bit/96 kHz converters. The AD/9624 accepts analog signals via balanced XLR connections and converts them to Lightpipe, four pairs of AES/EBU outputs, or TDIF (optional). The DA/9624 takes any of these digital data formats and delivers it as analog signals.

Sonorus's AD/24 ($799) and DA/24 ($599) are also 24-bit converters, but both are designed to work with digital data only in the Lightpipe format. Analog connections are handled by balanced XLR jacks, and the internal clock runs at 44.1 or 48 kHz. All three units provide word-clock connections.

Soundscape SS8IO-1. The SS8IO-1 ($1,695) provides a means of converting signals between TDIF and Lightpipe, and between either format and analog. The 2-rackspace unit offers eight channels of balanced analog I/O on XLR connectors, which are fed to 20-bit Crystal converters. Simultaneous transfer in both directions is possible. If you're connecting the system to a Soundscape DAW or audio card, the TDIF port is used to make the connection, so the SS8IO-1 can serve as either an analog or ADAT front end. Word-clock and Superclock RCA jacks are included. The internal sampling rate is switchable between 44.1 and 48 kHz.

Soundscape SS8IO-2. The SS8IO-2 ($349.95) offers many of the same functions as the SS8IO-1, except it lacks analog I/O. The half-rackspace SS8IO-2 can serve as a TDIF-to-Lightpipe converter or as an ADAT interface for a Soundscape mixing system. (If you're using the Mixtreme audio card, which has two TDIF ports, you could employ the SS8IO-2 as a Lightpipe I/O and use the spare TDIF port for TDIF, giving your system eight channels of each format.) As with the SS8IO-1, data can be transferred in both directions simultaneously, word-clock and Superclock connections are present, and the clock is switchable.

Soundscape SS8IO-3. The third offering from the folks at Soundscape is the SS8IO-3 ($599), which converts between eight channels of unbalanced analog audio and TDIF. This unit is designed to provide an inexpensive analog I/O solution for owners of the Mixtreme digital audio card, but it also serves as a stand-alone unit for converting TDIF to unbalanced analog.

Spectral Translator. The Translator ($995) gives you an effortless way of transferring eight audio channels from one digital format to another. The unit provides compatibility with Lightpipe, TDIF, SMDAI, and Yamaha Y2 devices. An upgraded model, the Translator Plus ($1,495), adds AES/EBU transfer.

The Translator can slave to external clock signals from any of the supported devices. If the selected source is not present, the Translator provides master clock to keep sync-dependent equipment running. A handy bypass mode ensures that you never have to disconnect cables.

Studer D19 MultiDAC. The MultiDAC ($3,175 for a basic system), part of Studer's D19 line of modular processors, delivers 24-bit D/A conversion from a number of formats. AES/EBU inputs are standard, and there are options for either ADAT Optical or TDIF inputs. Outputs appear on balanced XLR jacks. A monitor output is also included, and you can selectively monitor certain channels or sum all eight channels into a rough submix. This one is a stretch for the personal studio, though: I don't know of too many that are equipped with a Studer D19 rack.

Tascam IF-88AE. The IF-88AE ($1,245) is a single-rackspace unit designed to bring AES/EBU capabilities to a DA-series multitrack system. It has four XLR jacks that accept AES/EBU signals, in addition to a D-sub jack for TDIF signals. Conversion can take place in either direction, so the IF-88AE makes a decent stand-alone converter for any TDIF or AES device.

Tascam IF-TAD. Lo and behold, Tascam blinked first! The IF-TAD ($199) is a converter designed to connect ADAT Optical and TDIF devices. It has LEDs to indicate data present for each format, as well as a BNC word-clock output. I was floored when I saw the IF-TAD in an Alesis third-party developer catalog. When Alesis and Tascam start advertising each other's products, you know that times have changed for the better.

PIPING HOT

Perhaps we're getting a little carried away with all this fuss over digital connectivity. With the wide array of options and accompanying confusion, the eternal cynic in me says, "Give me an analog cable...I want a TT patch bay!" But I know those days are numbered, and that's probably for the better.

The death knell of analog has sounded, and the all-digital world is upon us. We must be prepared for the new millennium, and ADAT Optical and TDIF will play a big role in ensuring that we are. Without a doubt, these two formats will be around long after the ADAT and DA-series decks disappear.

MOTU's 2408 system includes a PCI card and this breakout box. The system supports 24 simultaneous channels, using any combination of its three 8-channel TDIF, three 8-channel ADAT Optical, two stereo S/PDIF, and eight analog I/O ports.

EM *Associate Editor* **Jeff Casey** *is still looking to get a record deal but recently realized that he probably needs to finish producing the record before that will happen.*

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creative guidelines for writing and producing children's music.

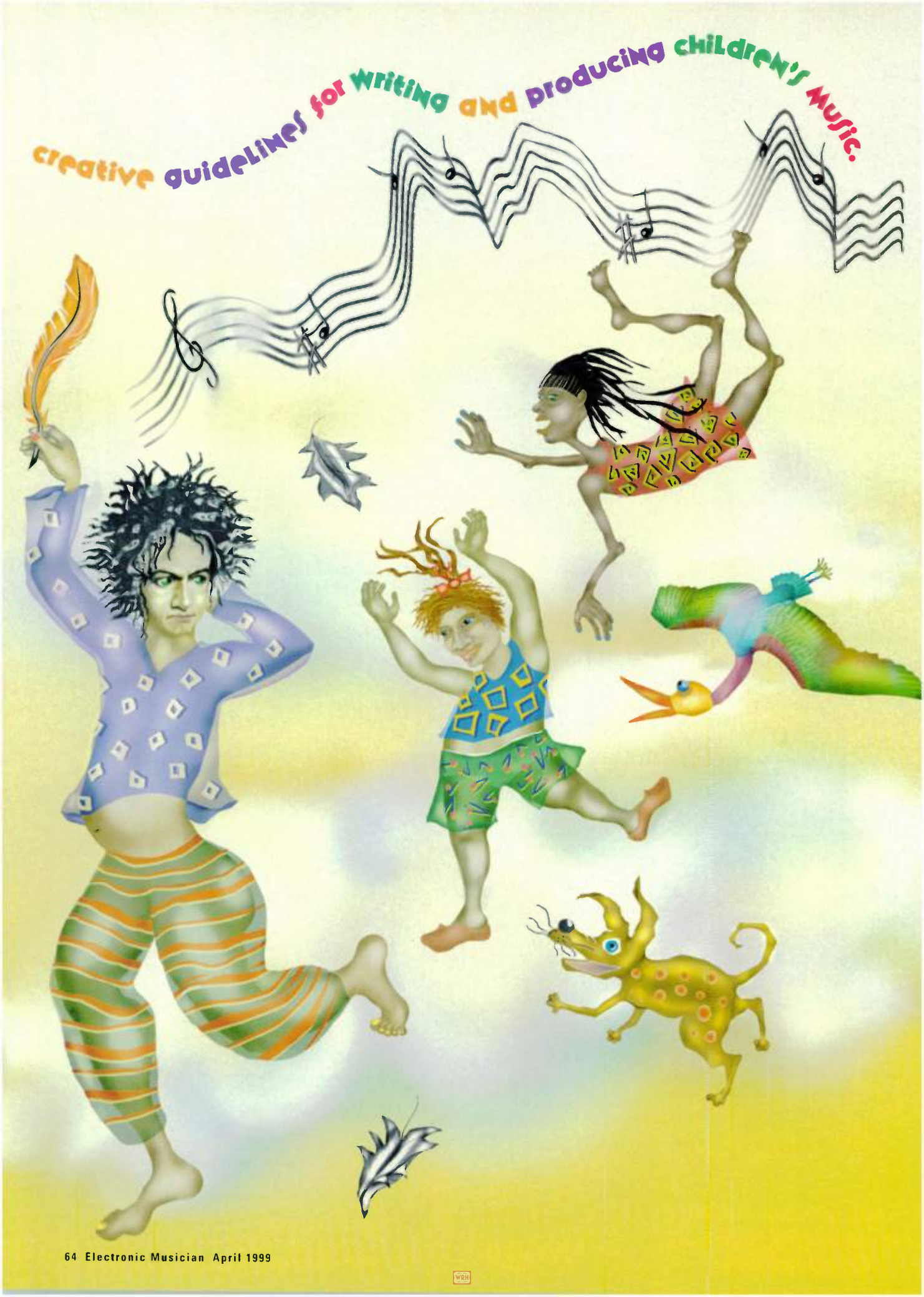

illustration by audrey colman

NOT JUST KIDSTUFF

The children's-music industry has been growing by leaps and bounds in recent years, and it shows no signs of letting up. Every week, it seems, a new artist—or an already established one—jumps on the bandwagon. Of course, not all those artists hit the mark. Some release great material but fail to get distribution, so no one hears their songs. Others get support from major labels but serve up pabulum. And even if a children's record sells millions thanks to blitzkrieg marketing, there's no guarantee that kids will actually enjoy listening to it. Parents buy the records, but the veto power belongs to the child.

BY BRIAN KNAVE

If you want to write children's music, or if you already write it and are looking for ways to improve your songs or your mixes, you've turned to the right page. This article is about what it takes to write songs that kids can't help but enjoy. It will cover essential elements of crafting songs for kids and relate helpful tips from some of the leading contemporary artists of the genre—Tom Chapin, Cathy Fink and Marcy Marxer, and the "godmother" of children's music, Ella Jenkins. (For information on selling your children's music, see "The Children's Music Market" in the July 1994 **EM**.)

Chapin, who is up for a Grammy this year for *In My Hometown* (Sony Wonder, 1998), has released seven children's records and a live-concert video. Fink and Marxer, whose recent release *Changing Channels* (Rounder Kids, 1998) has also been nominated for a Grammy, have nine children's albums and four videos to their credit. Jenkins, now with Smithsonian Folkways, is celebrating her 43rd year as a recording artist for children, having recorded 27 children's albums and two videos.

I'll also share insights gleaned from my own experiences writing, producing, and performing children's music. (I and my partner Norm Milstein, a well-known children's performer and all-around bard in the greater Sacramento area, wrote and produced a popular children's adventure/musical comedy called *The Search for the Moptop of Floppery Hair*. I have produced a number of other local children's artists and am currently recording Zapana, a husband-and-wife team that writes and performs South American–style children's songs sung in Spanish.)

Also included is a special sidebar about composing music to accompany storytelling, written by longtime **EM** contributor Paul Lehrman. Lehrman, who has produced music for more than a dozen children's recordings, discusses the tracks he recorded for the latest release of Sharon Kennedy, who is also his wife. Entitled *The Patchwork Quilt and Other Stories from Around the World* (Rounder Kids, 1998), it, too, is a Grammy nominee, though in the children's spoken-word rather than music category.

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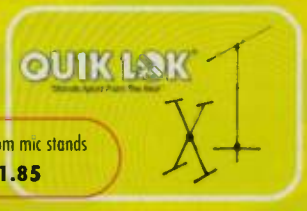

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KNOW YOUR AUDIENCE

Before getting started, it helps to determine which age group you're writing for. Basically, there are three: toddlers (ages 2 to 4), early elementary school (ages 5 to 7), and preteens (ages 8 to 12). "Raffi is the king of toddler music," says Chapin, "and I'm for Raffi graduates. After kids outgrow my music, they're off into radio."

Every child has unique sensibilities, of course, so the "crossover points" from toddler to early elementary, early elementary to preteen, and preteen to what could be called "the rebellious years" (when, by definition, kids embrace music their parents don't "get") can vary greatly, allowing for considerable overlap between the categories—and also allowing for certain titles to capture more than one audience. As Fink and Marxer point out, it's not uncommon for younger kids to listen "up." Older kids, however, will rarely listen to something below their level. Just the same, though, it's best to decide which audience you are writing for and then strive to make the songs more appealing to that age group.

ALL FOR ONE OR ONE FOR ALL

Kids can get obsessive when they really enjoy something. A well-loved record, for example, may get played over and over for weeks or months—a fact that parents can find excruciating, depending on the music. That's why some people also distinguish between *family music* and *baby-sitter music*.

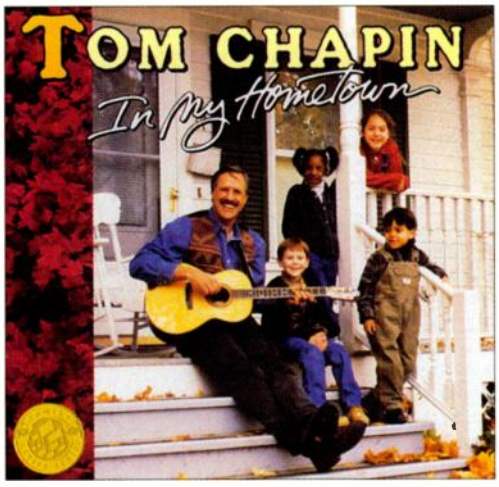
Family music is music the whole family can enjoy. Chapin, who is billed as a "family artist" by Sony Wonder, says he considers it the highest compliment when parents tell him, "Tom, yours are the tapes that we play on long car trips." Fink and Marxer, too, make clear their concern for engaging the whole family. "We want kids and parents to listen to our music together," says Fink. "We don't want our tapes to be designated as baby-sitters, the in-your-own-room-with-the-doors-shut kind of thing."

Woody Guthrie, who penned some great children's songs, was another advocate of family music—but with a somewhat different emphasis. "I want to see you join right in," he told parents at his concerts. "Let your kids teach you how to play and how to act these songs out. Get your whole family into the fun. I don't want kids to be grown-ups—I want to see the grown folks be kids."

While creating family music is perhaps the loftier goal, there's nothing inherently wrong with baby-sitter music. Barney, for example, is not most parents' cup of tea, but that hasn't fazed legions of toddler fans. Witness, too, the popularity of "personalized" recordings for kids, those on which a specific child's name is overdubbed to order into the songs. Sometimes the music on these tapes is annoying or just plain bad—but that doesn't stop children from enjoying hearing their names at every refrain.

Of course, it's possible for some titles to straddle the family and baby-sitter categories. One of our goals with *The Search for the Moptop of Floppery Hair* was to create something so "relatable" to kids that parents—particularly the super straight ones—would have a hard time getting the lyrics. Many adults have neither the time nor the patience for what reviewer Moira McCormick called the "gonzo stream-of-consciousness style" that we used on our record. But kids, once you get them engaged, will hang on your every word. We knew we had hit the mark when we started hearing parents say things like, "We love the music—but I didn't really understand the story until my three-year-old explained it to me."

One of the best-known artists for older children, Tom Chapin has released seven highly acclaimed albums and been nominated twice for Grammy Awards. His music, he says, is for "Raffi graduates."



Tom Chapin's latest release, *In My Hometown*, presents 15 original songs in a variety of musical styles—something one can pull off only in the children's music market.

STRANGE ATTRACTORS

So what makes a song a children's song? Interestingly, the subject matter has less to do with it than you might think. When I was a kid in rural Appalachia, there wasn't much children's music to speak of, so we listened to radio and records. The first two songs I remember latching onto (I loved them equally) were "Dang Me" and "Chug-A-Lug," both by Roger Miller. "Dang Me" is about a fellow so down on himself—and his luck—that he figures "they oughta take a rope and hang me." And "Chug-A-Lug" recounts a man's fond memories of sneaking off to get drunk when he was "just going on fifteen."

Clearly, neither of these songs is what most folks would consider children's music. Children, however, played a decisive role in the success of "Dang Me." Initially, a song called "Less and Less" was slated as the single for Miller's debut album. But one night, while producer Jerry Kennedy was reviewing session tapes at his home, his children happened to hear "Dang Me" from their bedroom. "My kids came screaming down the stairs when the song came on," says Kennedy. "They thought that was the greatest thing they'd ever heard." Due largely to their

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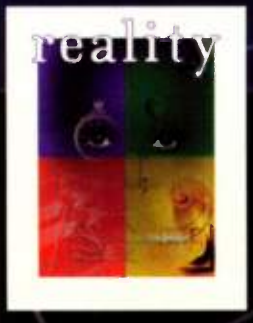

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Gregory Butler is a producer/writer/remixer. From Alternative to Hip Hop, for records and movies, **Reality** is the synth that gives Gregory the power and versatility to create his music.

"When writing or remixing, I always start with sound. Crafting the sounds beforehand is the most important thing I do. I get deep inside the **Reality** engine to create the sound and let the sound and the mood of what I create write the part. I look for a piece of equipment or software that allows me to think of how I want to work as opposed to forcing me to work in a specific way. That's what it gives me.

"Designing sounds with **Reality** is very straightforward. I can be a lot more creative and experimental looking at a 29-inch monitor than I can on a half-inch LCD. If I want to play with the LFOs, I can see what's going on. I can switch things randomly or I can be very precise, depending on my mood. The engine is incredibly flexible.

"When you're doing this for a living, you look at a piece of equipment and ask 'Is this going to change my life? Is it going to make it easier? Make my songs better? Is it going to make more money for me than I am spending on it?' Then I ask myself, 'Does this do something that **Reality** doesn't do?' and it never does.

"Also, when it comes time to sell a synth, you've lost the sounds. With **Reality**, I am not likely to sell it, because it's simply a program—it's always going to be there. It simply takes up a little room on my PC."

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"With **Reality**, you can go on forever."

Gregory's credits can be found on several movies and many albums and remixes. Just look for some of Gregory's pseudonyms, like **the enemies**, **steve zodiac**, and **dcoy**. Recent credits include **K's Choice**, **Lacksidayze** and **Switchblade Symphony**.

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enthusiasm, Kennedy persuaded the label to ditch "Less and Less" and to release "Dang Me" in its place.

Chapin considers clarity, not content, one of the most important qualities of a good children's song. "For an adult record," he says, "you can do, say, a tone poem that almost doesn't mean anything, or a song that's very spacey lyrically, or something very poetic. But kids aren't going to sit for that. It gets boring for them. Not only boring, but upsetting. They want to know: What is this about? What does it mean? And you lose them. So the key is clarity. Lyrics don't have to be simple—children can handle very sophisticated ideas and lyrics, if you do it right—but they do have to be clear in the sense that kids know what you're talking about and can see the pictures. You can even use words they don't know—but you can't have a situation in which they get frustrated because they don't know what's going on."

Fink and Marxer also feel that subject matter for kids is fairly open-ended, as long as songs aren't frightening or depressing. "At an adult gig," points out Marxer, "you can sing the saddest songs in the world, and that's fine. But adults have reasoning powers beyond what children have and can say either, 'I'm going to go there' or 'I'm not going to go there.' But kids *will* go there. I think it would be irresponsible to do a song that's a real downer and then just leave the kids down. You can tackle pretty much any subject, though, as long as you use humor, keep your heart open, and keep perspective on things."

Cathy Fink and Marcy Marxer—better known to their fans as simply Cathy & Marcy—not only are highly accomplished singer/songwriters and producers of children's music, but they're also first-rate musicians.

ELEMENTS OF APPEAL

There's no formula for writing a children's song, but there are techniques to help ensure that kids will be engaged and, hopefully, delighted by what you've written. The key word here is *delight*. As Chapin points out, "the only rules are to make it delightful and full of ideas and pictures."

***Changing Channels* by Cathy Fink and Marcy Marxer is a program of 12 original songs intended to "help kids process their TV and media influences in a positive way."**

Literary techniques such as rhyme, repetition, and alliteration can increase the appeal of most any song. That's because they make songs easier to memorize and more fun to sing. One of the "hit" songs on *The Search for the Moptop of Floppery Hair* is "Family Reunion." Check out the alliteration and rhyme in this verse:

When the bumblebees are buzzin'
And the basset hound's a barkin'
And the barbecue's a bubblin' in the backyard
And Kate your kissin' cousin
Is kissin' cousins by the dozen
And your mama's slurpin' pickles from a jar

Humans learn through repetition, and kids have a natural affinity for it. Repetition, aside from driving home a particular point, is also enjoyable. In the song "Ha-Ha This-a-Way," Huddie Ledbetter uses repetition liberally in the verses and the chorus:

When I was a little boy, little boy, little boy
When I was a little boy twelve years old
Papa went and left me, left me, left me
Papa went and left me, so I was told
Ha-Ha this-a-way, Ha-Ha that-a-way
Ha-Ha this-a-way, then, oh then

CONCRETE FEAT

Another element that good literature and good songwriting share is the use of concrete language, or words that denote specific things, actions, people, and places. Concrete language calls up immediate, and similar, images for everyone. Abstract language, on the other hand, rarely evokes images, and if it does, those images differ from one person to the next. For example, if I say the word "concrete," everyone sees a similar image—the grayish white stuff that hardens into sidewalks and foundations. But if I say "abstract," what do you see? Not much (unless you're the scholastic type and the word calls up images of papers). For the most part, then, *abstract* is an abstract word.

Here are some other abstract words: *freedom, peace, change, love, bliss, danger.* In one sense, we all know what these words mean, but each one usually means something different to everyone. More importantly, abstract words do not paint pictures in the mind. What does freedom look like? Or bliss? Although people may have images associated with these concepts, the good writer knows not to rely on those associations. They are simply too arbitrary.

Concrete language, however, is specific and can be relied upon to perform consistent image making in listeners' minds. (Note the number of concrete words in the verse above from "Family Reunion.") Comb your lyrics in search of abstract words. For any you find, try to think of a more concrete way to say the same thing. Although it's not always possible (or appropriate) to replace an abstract word with a concrete one, when artfully done, it almost always improves the immediacy and intelligibility of the lyric.

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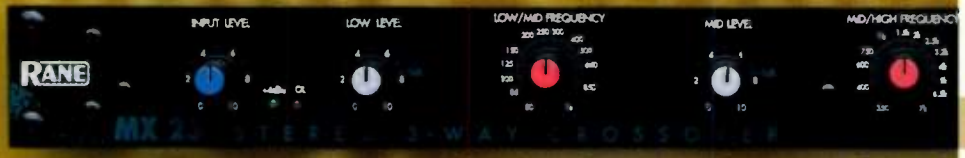

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HUMOR US

It's no accident that half the children's songs in *Rise Up Singing*, the classic group-singing songbook, are listed under the category "Play." (The other half are listed under "Lullabies.") Children love to play and to laugh. Therefore, anything you can do to make your songs funny will greatly increase their appeal to children. Of course, humor depends a good deal on delivery, but it helps if the material itself is funny. (The only caveat there is that what's funny to an adult may not be funny to a child—and vice versa.)

At the word and sentence levels, humor can often be accentuated by choosing fun and colorful words (for example, *galumph*, *slapdash*, *skedaddle*, *lickety-split*) over ordinary ones (*prance*, *hasty*, *leave*, *headlong*), or by using ordinary words in funny or novel ways (for example, "An omelet is egg-zactly what I'd like").

NO PRETENSE

Children are straight shooters. For example, if you have a wart on your nose, the average kid will ask you what it is and how it got there. If you have a lisp, he'll ask, "Why do you talk so funny?"

MUSIC FOR STORYTELLING

"The music is not front and center—it's there to enhance the story," says Steve Netsky of Rounder Records, coproducer of the Grammy-nominated album *The Patchwork Quilt and Other Stories from Around the World* by Sharon Kennedy. "If someone listened to the album and said later, 'Oh, was there music?' that would be okay with me. Sometimes you do want the music to take center stage: for example during interludes or transitions between stories. But even though the music is not the focus, kids listen to these stories over and over, and after a few listenings they'll start paying more attention to the music. Therefore, you want it to be exactly right."

Storytelling may be the oldest form of human communication, but it doesn't occupy much space in the catalogs of major record labels. Professional storytellers—as differentiated from children's entertainers, book readers, or actors performing material for children—traditionally produce their own recordings and sell them at gigs, or occasionally through catalogs. Rounder took a chance a couple of years ago and signed Kennedy. Her first album, *Irish Folktales for Children*, did well enough to encourage the company to do a second.

Storyteller Sharon Kennedy's *The Patchwork Quilt and Other Stories from around the World* is a multicultural collection of stories.

Music plays an important part on Kennedy's records. The Irish album, for example, makes use of traditional instruments such as harp, fiddle, accordion, and bodhran. For the second, more eclectic album, the production team of Netsky, Kennedy, and Bing Broderick decided the stories should be accompanied by a more diverse selection of instruments, including guitar, banjo, piano, recorder, drums, hand percussion, bagpipes, and male and female singers. I got to play many of these instruments, because I have an "in"—I'm married to Sharon Kennedy.

"If it's not logistically impossible," says Netsky, "I try to have the storyteller and the musician play simultaneously, so that their performances can inspire each other. But sometimes, that just doesn't work." In a story about a Scottish sea serpent, for example, the producers wanted to intersperse a mournful bagpipe track. But even with Kennedy in an isolation booth, there was so much leakage from the bagpipes that there was no way the tracks could be recorded at the same time. Instead, the bagpipes were recorded first as one continuous track, the player piping an air for a length of time equal to the length of the story. Then Kennedy recorded her part while engineer Eric Kilburn, following a script, brought the pipes up and down in her headphones at prearranged times so she could get the benefit of the music. The process was repeated almost exactly during the mix, and the track and the story fit together perfectly. "We decided that everyone should have a bagpipe track parallel to their lives," jokes Netsky.

Microphone choice and technique are critical to making a good storytelling tape. "I'll try several mics on a storyteller," says Netsky. "Then, we'll all compare the tracks carefully until we agree on which tone we like." Again, because these records are listened to over and over, the slightest glitch or harshness can become a major annoyance. Storytellers are different from voice-over artists and even stage actors in that they are used to having to project to large audiences. Therefore, their vocal and physical gestures tend to be much more pronounced. And because there is often no other sound being made while they are speaking, special care must be taken to keep extraneous noises such as mouth smacks and tongue clicks off the track. True, unwanted sounds can largely be eliminated by digital editors, but that approach can get expensive. It's best to get the sound right from the get-go, using proper mic selection and placement to prevent sonic gremlins. For example, switching to a less sensitive mic might help, or moving the microphone a bit farther from the artist than you would otherwise. Monitoring at relatively high levels can also help ensure a clean track.

—Paul Lehrman

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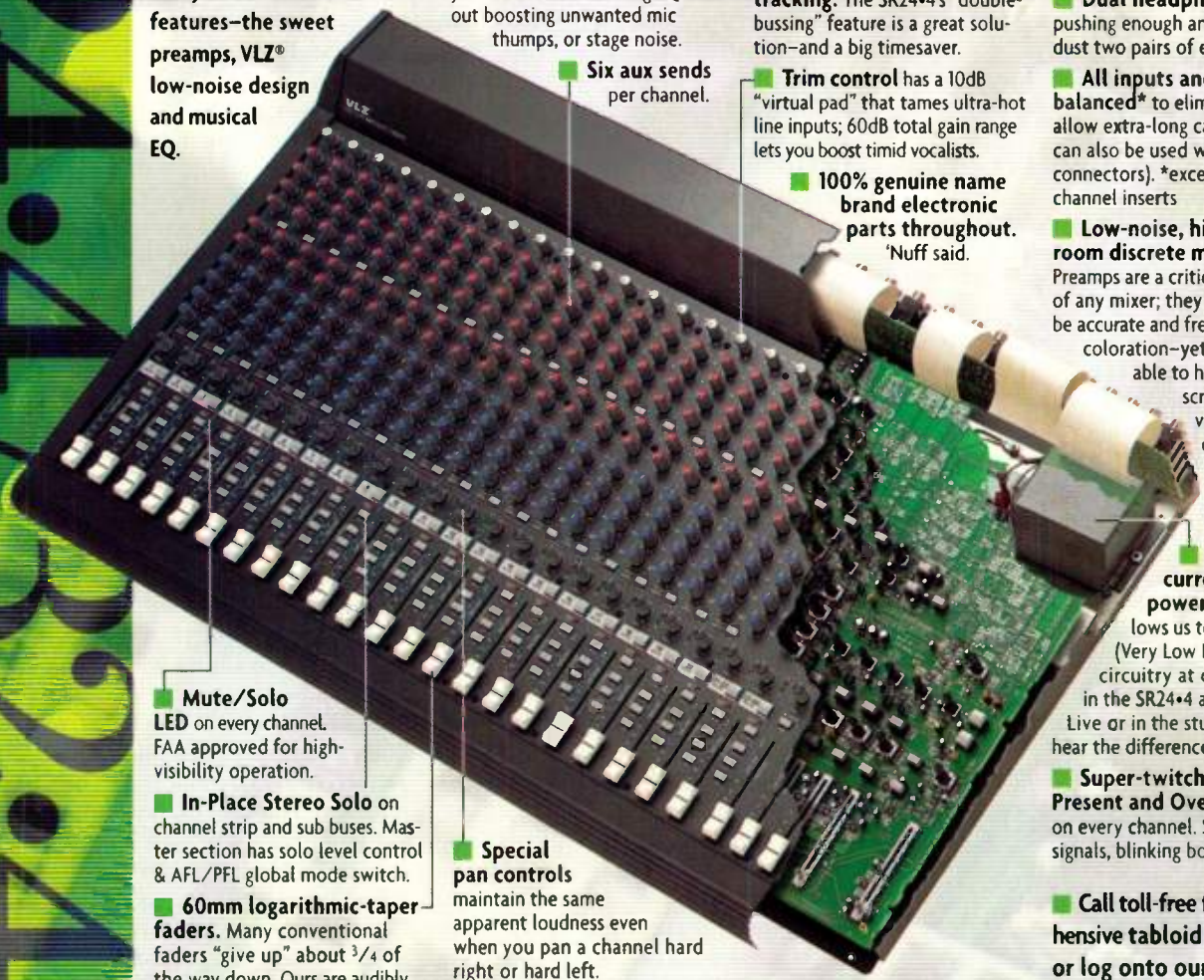

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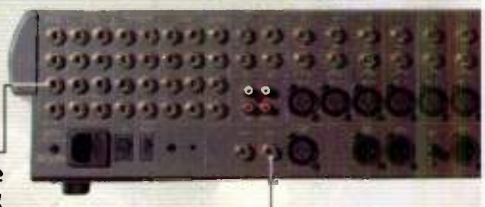

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Understanding their candid curiosity is key to writing for children.

Check out these verses from "Why, Oh Why" by Woody Guthrie:

> Why can't a bird eat an elephant?
> Why, oh why, oh why?
> 'Cause an elephant's got a pretty hard skin.
> Goodbye, goodbye, goodbye.
>
> Why can't a mouse eat a streetcar?
> Why, oh why, oh why?
> 'Cause a mouse's stomach could never get
> big enough to hold a streetcar.
> Goodbye, goodbye, goodbye.

Clearly, there are other, more obvious reasons that a bird can't eat an elephant and a mouse can't eat a streetcar. But those other reasons don't matter to a child. Left to their own devices, kids aren't terribly concerned about correctness, appropriateness, or decorum—they'd much rather play, create, and have fun.

In the above verses, the illogic of the answers is what makes them appealing. Kids have a penchant for illogic (at least until we drum it out of them) and their fascination for things bizarre, stupendous, and nonsensical knows no bounds. Think *Alice in Wonderland* and Dr. Seuss. Those formats are the ones that get most kids off—not dull, overly rational, left-brain constructions.

Likewise, children also enjoy some mischief in songs and stories. (It's not for its political correctness that *Grimm's Fairy Tales* has so long remained a children's classic.) Chapin calls this element anarchy. "I'm very clear on the fact that children's songs need to have a little bit of anarchy in them," he says. "You know, where someone's being a bit naughty. Kids can relate to that."

The author (right) with his children's-music partner, Norm Milstein. Together, they are known as Norm & Brian—and yes, that's a spatula.

FOUR SONG TYPES

Certain types of songs, when well written, are engaging for most children most of the time. (If you've ever played a children's concert, you know that nothing works *all* the time.) I've listed four tried-and-true song types below. Keep in mind that the categories aren't strict, and there's plenty of room for overlap.

Activity songs. Kids are energetic and they love to participate. Therefore, do whatever you can to involve them physically in your songs. As a child, my favorite Sunday school song was "I'm in the Lord's Army" because it allowed us to act out the lines "march in the infantry/shoot off artillery/fight off the calvary." Although in retrospect I can see that it was a fairly inane—even insane—song, that hardly mattered; what mattered was that we got to stomp and shoot and brandish imaginary swords—in church, no less.

Singing, of course, is a primary activity for kids when they listen to music. With regard to form, there are numerous options to choose from, including sing-alongs, rounds, and call and response. (Ella Jenkins's first record, released in 1956, was entitled *Call and Response.*) Children also love to make animal noises and other "mimickable" sounds. When I perform "Family Reunion," for example, kids in the audience enjoy making the sounds that occur in each line: a bumblebee buzzing, a basset hound barking, a mother "slurpin' pickles from a jar."

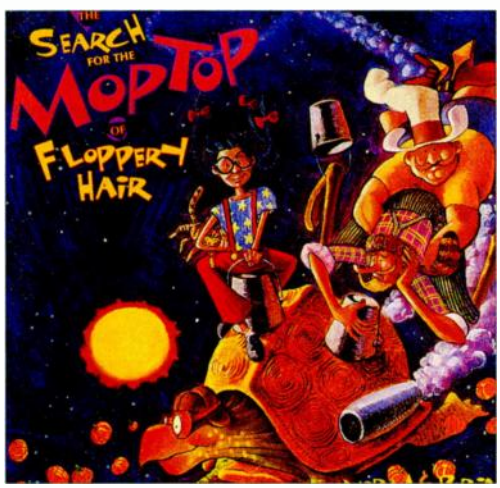
Dancing is another favorite activity. Remember the Macarena? Kids of all ages—including my niece, who was three at the time—learned the steps and hand motions to that one. Come up with an original dance to one of your kids' songs, and you'll enhance its appeal tenfold.

Story songs. One of the best recipes for a children's song is a story. Kids, especially little ones, love tall tales, fables, and fantasies (as Disney well knows). Other types of stories can be worked into songs, as well. Just be sure to choose colorful characters with fun, memorable names. Dozens of outlandish characters populate *The Search for the Moptop of Floppery Hair,* including the Woggaloop (a salsa-powered flying tortoise), Zorky Poon (a blues harmonica player from the south side of Neptune), and Harold the Intergalactic Circus Cow, who specializes in tightrope walking.

Older kids may prefer more realistic story songs, especially those that relate to their own experiences. On *Changing Channels,* Fink and Marxer have a song called "Buy Me This and Buy Me That" in which a family goes to the store and the child, excited by several toys, asks, "Hey, Dad, can you buy me all that stuff?" Later, he sees some more toys on TV, and he asks his mom for those. But both times he gets an answer (delivered as the refrain) that most any American kid can relate to:

> "It's buy me this and buy me that
> Buy me that and buy me this."
> She [Mom] smiled at me and blew a kiss.
> "Do you think money grows on trees?
> If so we'll have to plant some seeds.
> You can't have everything you want
> And don't forget to say please!"

The plot takes a turn in the third verse, though, when Dad sees a new bike he wants. Mom answers, "Oh, honey, if



***The Search for the Moptop of Floppery Hair* by Norm & Brian combines music, poetry, humor, and storytelling. According to reviewer Moira McCormick, "This resoundingly original story unspools in a gonzo stream-of-consciousness style."**

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you buy that bike today/Then I want fancy new rollerblades/Knee pads, wristguards and cool-looking shades." The child can hardly believe his ears. Now it's *his* turn to sing the refrain.

Wordplay. Most children relish words, so spare no effort to be playful and clever with language in your lyrics. Wordplay, puns, colloquialisms, bizarre locutions, unexpected variations and juxtapositions—there are any number of ways to spice up your lyrics and make them more fun for kids. As Lewis Carroll demonstrates in "The Jabberwocky," even meaningless neologisms can have immense appeal.

Chapin has a fun wordplay song entitled "Uh Oh, Accident" in which the word "accident" undergoes various rhymed—and curiously logical—transformations:

Fell off my bike and it got bent.
Uh oh, accident.
Oh no! Bike bent. Accident!

Picked up a hammer and I hit my thumb.
Uh oh, accidumb.
Oh no! Poor thumb. Accidumb!

Took me a bath and I ate the soap.
Uh oh, accidope.
Oh no! Ate soap. Accidope!

Milstein employs a similar technique in "Macaroni and Cheese," another song from *The Search for the Moptop of Floppery Hair:*

When I'm down and I'm all aloney
When I'm broke and I haven't got any moaney
I don't give out a gasp or a groaney
And I don't eat any sandwich baloney

I break out the macaroni and cheese
The macaroni and cheese

If you're Portuguese or Italeeony
Whether you're from Utah or Arizony
If you're big and brawny or kinda bony
If you're a soprano or a baritony

Just break out the macaroni and cheese
The macaroni and cheese

KIDS PICK THEIR THREE FAVORITES

Kids' tastes are anything but predictable. To get a sense of the range of children's musical preferences, I had some of the parents around the office ask their children to select their three favorite music albums. The results were all over the map. (By the way, the kids who chose my title, *The Search for the Moptop of Floppery Hair*, did so without any prompting. Of course, I do have the advantage of knowing their parents!)
—Brian Knave

Isabelle Forlin,
age 3
***Children's Music Collection,* various artists (Smithsonian Folkways, 1998);**
***My Songs* (personalized songs from Baby Stuff);**
***Not For Kids Only,* David Grisman/Jerry Garcia (Acoustic Disc, 1994)**

Shelly Gleason,
age 5
***Choo Choo Boogaloo,* Buckwheat Zydeco (Music for Little People, 1994);**
***On the Sunny Side,* Maria Muldaur and Friends (Music for Little People, 1990);**
***The Search For the Moptop of Floppery Hair,* Norm & Brian (Moptop Records, 1995)**

Drew Jackson,
age 3
***Silly Time Magic,* Joanie Bartels (Discovery Music/BMG Kidz);**
***The Best of the Gipsy Kings,* Gipsy Kings (Nonesuch Records, 1993);**
***Children's Favorite Songs* (Walt Disney Records, 1979)**

Kelly Lawson,
age 5
***Golden Hits,* Sarah Vaughan (Mercury Records,1990); *Space Jam* (movie soundtrack, Atlantic Records, 1997); *Jefferson Airplane Loves You,* Jefferson Airplane (RCA Records, 1992)**

Jesse and Jake Pledger,
ages 4 and 6
***Gift of the Tortoise,* Ladysmith Black Mambazo (Music for Little People, 1994); *Jump Up,* Chris Molla (Fingerpaint Music, 1997);**
***All For Freedom,* Sweet Honey in the Rock (Music for Little People, 1989)**

Zoe Semler
age 6
***Mulan* (movie soundtrack, Walt Disney Records, 1998);**
***China, A Time to Listen* (Ellipsis Arts, 1998);**
***The Search For the Moptop of Floppery Hair,* Norm & Brian (Moptop Records, 1995)**

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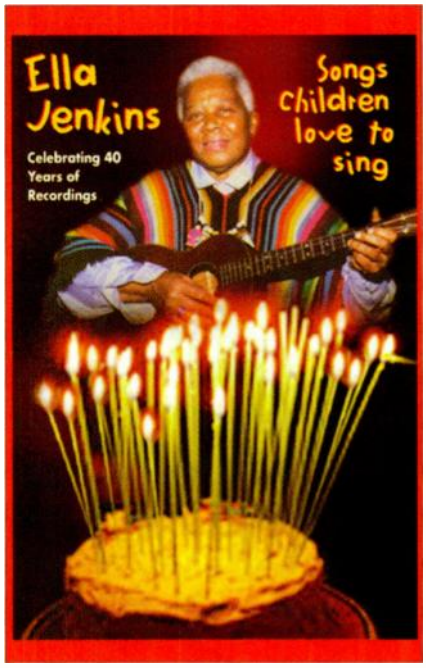
A common misapprehension is that little kids can't handle big words. But my experience has been that even preschoolers can get turned on to multisyllabic words, especially when the words are fun to say aloud. (Consider the lasting popularity of the song "Supercalifragilisticexpialidocious.") It doesn't matter if the kid knows what the word means or not—she can always learn, which is part of the fun.

Listing or "cumulative" songs. *Listing* is when you add a new line to the refrain each time it comes around. This is a fun challenge for kids because the list gets longer each time. A good example of a listing song is "The Twelve Days of Christmas."

On *Changing Channels*, Marxer has a song that combines listing with activity. Here's the penultimate verse, which contains the completed list, of "A Ballet Dancing Truck Driver." Marxer sings the lines, and the children yell out the parts to the right of the dashes:

I looked up at the moon
And now I want to be
An astronaut—LIFT OFF!
A math teacher—2 and 2 is 4
A carpenter nailing—tap, tap, tap
The finest cook—Soup's on!
An elephant tamer—Sit here, please!
A runner racing—READY, SET, GO!
And a ballet dancing truck driver
(A ballet dancing truck driver)

Other types of listing include alphabet songs or any subject matter organized by category (numbers, colors, types of clouds, and so on). Many of Jenkins's songs use a similar technique of setting up a category that allows for variations suggested by children in the audience. For example, in the call-and-response classic *"Pole Pole"* (Swahili for "go slowly, go slowly," pronounced "poh'-ley poh'-ley"), Jenkins first sings about several African animals, then she encourages the kids to call out other animals to insert into the song.



Songs Children Love to Sing **is an anthology celebrating Ella Jenkins's 40 years of recording for Folkways Records and Smithsonian Folkways. It includes 17 of Jenkins's most-requested songs.**

HARD LESSON TO LEARN

Unlike most pop music, which is intended primarily for entertainment (and sometimes also for inspiration), children's music often has a third purpose, which is instruction. For example, children's music can introduce concepts (a song about sharing), people (a song about an important historical figure such as Martin Luther King), and places (a song about Antarctica). Songs are also a great way to teach children new words and languages.

For example, Jenkins has a song entitled, "I Know a City Called..." in which children complete the refrain by yelling out the name of a city. She got the idea while traveling on a highway in Florida. "I saw this big sign that said 'Okeechobee,'" relates Jenkins, "and I thought, 'Oh, my, what a great name!' I'm fascinated by long and unusual names, and children are, too. They love to experiment with new sounds and new words. Even if they're singing only the last word of the refrain, it introduces them to a new word and a new place." Like "Pole Pole," "I Know a City Called..." is open-ended, allowing children to bring their own experiences to the song during performances—in this case, supplying the names of other cities they have been to or heard of.

There are other ways that music instructs, of course. Even in songs written purely for entertainment, the use of real instruments alone, played authentically and in tune, is a form of musical instruction. The children's artists I interviewed for this piece are all known for their use of real instruments. For Fink and Marxer, it's a critical issue. "We have developed a very hard-core philosophy through the years," says Fink, "that part of what we're doing is giving kids a chance to hear real music and real musicians. So we don't use synths and we don't fake anything. Instead, we bring real musicians into the studio. Of course, from time to time we've relied on people who are expert at MIDI or synths, but only when it's appropriate to the style. If we're recording a rap song, for example, MIDI production is part of it. But if it's a Dixieland tune, we're not going to use a bunch of horn patches. We never use drum machines, either, and we strive to record everything well so that kids get to hear the real thing."

Instruction is also inherent in different musical styles, which can provide "lessons" in culture and geography. In *The Search for the Moptop of Floppery Hair*, for example, the main characters go in search of a man's stolen hair and encounter a different style of music each place they land. They hear a blues song in Chicago, for instance, a country song in Nashville, and a reggae tune in Jamaica.

CONCEALING THE PULPIT

One of the more difficult things to write for children is an instructional song that conveys a complex or abstract concept. The trick is to avoid preachiness in favor of telling a good story—one that's clear enough for the child to follow whether he gets the message or not.

"One of the things that drives me nuts," says Chapin, "are the kids' records that are so 'PC' and bland, with no sense of humor. The parent thinks, 'This is good medicine.' But what you need with a kids' record is good medicine that doesn't taste bad. This is not church and it's not school—it's entertainment. So never say 'should.' If you want to write a song about recycling, for example, don't say, 'You should recycle because it's good for the environment.' Instead, tell a story that's entertaining but that makes the point, as well."

When Chapin and cowriter Michael Mark took on the subject of recycling, they came upon the idea of telling the story in terms of a dinosaur.

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A stegosaurus in the forest munching on
some hay,
Lay down to snooze in a bed of ooze and sadly
passed away.
Her body changed and rearranged as she
sank beneath the soil,
And over time she turned to slime and then
she turned to oil.

So she disappears for a jillion years 'til
finally she is found
By a guy named Bill with an oil drill who sucks
her from the ground.
Bill pipes that poor old dinosaur to an oil
refinery,
Where they bubble and boil and change that
oil into stuff for you and me.

R-E-C-Y-C-L-E.
That's the way it's s'posed to be.
The earth recycles and so do we.
R-E-C-Y-C-L-E.

The song goes on to describe how petroluem from the Jurassic era is used to make plastic jugs, mugs, drinking straws, and so on. The song's only words of instruction, though, come from the dinosaur—a source children are more likely to heed than a preachy adult.

And if a dinosaur could see us pouring
orange juice this way
From a plastic jug we made of her, well this is
what she'd say:
"It took me 80 million years to get to you
today.
So don't throw that mug, don't throw that jug,
don't you throw me away."

"When kids first hear this song," explains Chapin, "their eyes open wide as they see the pictures, follow the story, and laugh at the internal jokes. But it might take until the fourth or fifth time they hear the song before the point starts getting across."

In another song entitled "Good Garbage," Chapin and cowriter John Forster use simple, memorable language to address the equally complex subject of biodegradability. Here's the chorus:

Good garbage breaks down as it goes.
That's why it smells bad to your nose.
Bad garbage grows and grows and grows.
Garbage is s'posed to decompose.

STYLE, STRUCTURE, AND MELODY

The world of "adult" music is style specific: if your material is too eclectic or doesn't fit a particular genre, the record companies won't know how to market you. But the great thing about children's music is that it isn't limited to a single style. As Marxer notes, "For a kids' record, the genre is kids." Therefore, when writing for children, it makes sense to explore as many styles as you can handle musically. Kids are easily distracted anyway, so a variety of styles and instruments can help hold their interest.

As for song structure, the basic ballad form, with verses that inform and lead up to a singable, hooky chorus, is usually your best bet, regardless of musical style. "That's the form that I tend to use the most," says Chapin, "and it really works great. There's nothing quite as infectious as a great chorus."

Although there's no prescription for writing great melodies, one thing I will say is, avoid those that sound like a well-meaning adult's idea of what a melody for children *should* sound like. You know what I mean: the cloying, corn-syrupy, goo-goo-ga-ga stuff that immediately announces itself as "children's music." Believe me, children of all ages can handle good, interesting, uninsipid melodies. (Kids love The Beatles.)

> "What you need with a kids' record is good medicine that doesn't taste bad."
> —Tom Chapin

As for arrangements, Chapin suggests tightening each song as much as possible—even if it means doing away with instrumental solos. "On one song, we had Branford Marsalis and he played this sax solo to die for," relates Chapin. "It killed me, but when we mixed the song we had to say, 'Hey, this is a kids' record.' So we cut the solo short. When we play live, we might put in a solo where there isn't one on the record, because we know it will fly. But on the records, it's not about the solos—it's about the stories and the tales."

Fink and Marxer, on the other hand, see the instrumentation as being more central to their musical purposes. "If we have a great jazz combo come in," says Marxer, "we might have them trade fours at some point in the song so that kids can hear each instrument clearly. Otherwise, how are children going to know if they want to grow up to be a bass player or a drummer or whatever?"

NOTES ON THE MIX

Most children's music is lyric driven, so be sure to keep the vocals up front and, generally, not too awash with reverb. "You can't mix a children's album like the early Springsteen or Stones records," says Chapin, "where you can barely hear the lyrics. Rather, it should be mixed more like modern country records, where the voice is right out there and you can really understand what's going on."

Fink and Marxer, who record many of their tracks at home but mix at a pro studio, are also very conscious of vocal levels. "We're always struggling with that fine line between making sure that the vocals are loud enough and that we haven't lost the energy of the band," says Fink. "We tend to mix the vocals a little more up front than we would for our adult songs, and it's trial and error to do that without losing the band energy. But if the vocals are buried in the mix, then you've lost the song."

THE ACID TEST

Before committing your children's songs to a recording, it's helpful to test them live. "If you aren't a children's performer by trade," says Fink, "then call up your local elementary school and find a few classrooms that will let you bring in your instruments and perform. Then you can see if your songs even work. See if they feel good, if they're fun for the kids, and if the kids sing along. A lot of people have an idea of what a kids' song is, and yet their songs don't work in real life. The songs need to be tested, both with kids and parents. And I don't mean your best friend's kids who are going to love anything you do no matter what it is—I mean a group of kids you've never met before, where you have to sell the song and see if it holds up or not."

COURTESY ELLA JENKINS

With nearly 30 titles in 40 years, veteran Ella Jenkins is perhaps the most widely recorded children's artist in the world. She is shown here with her beloved tenor (four-string) guitar.

Marxer recommends an even more challenging environment. "Volunteer at a hospital," she says, "a place where kids aren't feeling well, where they're a little bit crabby and there are lots of interruptions. If you can grab kids with your material there, you're on the right track."

Jenkins, too, tests new songs on kids before adding them to her repertoire. "If I don't get a really positive response," she says, "I'll just put the song on the back burner for a while." She suggests checking out other children's performers to "see what kind of appeal they have, how they are able to hold the children's attention. Not that you're going to copy them, but just to get inspired."

CHILD MIND, BEGINNER'S MIND

Naturally, the music we write for kids is shaped by our beliefs about children—who they are, what they want or need, and how best to present it to them. A person who believes that children should be seen and not heard, for example, will write a very different children's song from someone who believes that children's perspectives and opinions are of value.

The best children's music, I believe, proceeds from a deep love and respect for kids, coupled with a genuine desire to entertain. "You really have to like children," says Jenkins. "You have to feel that they have value and worth, and that you have something valuable to share."

Most importantly, children's music should not be approached as being inferior to other music. "There are a lot of people who oversimplify what this is about because it's for children," says Fink. "I want to warn newcomers that they can't get away with less just because it's for kids. If anything, we feel that we need to be giving them more." Chapin agrees. "In many ways, it's harder than regular music," he says, "because you're dealing with two audiences: you're trying to make songs that please you and that please kids. And that's much harder than you think it is. Anyone who comes in here saying, 'I can't make it as an adult performer so I'll just be a kids' performer, instead,' is in for a rude surprise. The fact is, there's an energy required to do this thing right. When it's for kids, it can never be too good."

Brian Knave *is an associate editor at* **EM**.
Thanks to Ella Jenkins, Tom Chapin, Paul Lehrman, Mary Miller, Cathy Fink, and Marcy Marxer (www.folkmusic.com/cathymarcy).

RECORDING MUSICIAN

Ten Mics I Swear By

In which the author presents his well-considered opinions.

By Myles Boisen

Here's a question that most recording engineers have heard at one time or another: "What's the best microphone?" A very insightful response to this question appeared on the rec.audio.pro Web site in the form of another question: "What kind of ice cream is best?" Some people like vanilla. I usually go for Cherry Garcia, but tomorrow I may be in the mood for chocolate. In other words, the best microphone is the one that sounds best on a particular instrument for a specific project and comes at a price you can afford.

Two of the better values on the market, the Oktava MC 012 (center) and MK 219 (right) cover a lot of tonal ground. The MC 012 is modular, providing an in-line 10 dB attenuation pad and three interchangeable capsules (cardioid, hypercardioid, and omnidirectional). It comes with a shock mount, too (left).

That said, I will now list ten of my favorite microphones, all but one of which cost less than $2,000 retail. This list is subjective, based on my experience as a recording engineer and a reviewer for **EM**. I regularly reach for each of these microphones from the collection of roughly 50 brands and models that are found in the Guerrilla Recording mic cabinet.

This list, originally compiled as a general post to the rec.audio.pro newsgroup, was intended to address some frequently asked questions, as well as to guide readers toward good all-around studio microphones. Of course, such lists are necessarily works in progress, because new products are released continually. It's quite possible that, of the many new microphones on the market that I have not worked with, some could well infiltrate these ranks and usurp a position or two. But at the same time, rest assured that most of the mics discussed here have withstood the test of time, and I expect that they will not lose their appeal anytime soon.

You will notice that certain "legendary" Neumann and AKG tube mics are not included here, nor will you find any mics that are particularly esoteric. The only reason for these omissions is

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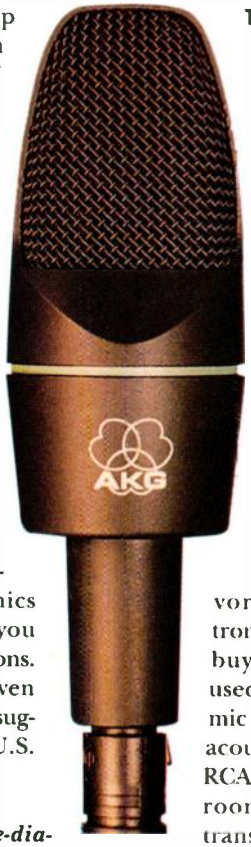
that I haven't yet worked up the courage to pay more than $2,000 for a single piece of gear. I like the more expensive toys, of course; I like to read about them, and I like to hear them used well on recordings. But I *love* great-sounding gear that I can actually afford to own.

Please note that I generally use these mics with class A or high-quality tube preamps, and I usually record to analog tape. Naturally, depending on the type of gear you use, the way you record, and your personal tastes, the results you get from these mics may differ from mine, or you may reach different conclusions. Also, note that the prices given here are the manufacturers' suggested retail prices for the U.S. market.



The AKG C3000 was always a great mic, but since AKG slashed its price in half a few years back, it has become an irresistible buy.

AKG C 414 B/ULS ***large-diaphragm, multiple-pattern condenser.*** The smooth response, extended highs and lows, and versatile features of this mic—combined with its durability, market longevity, and accessible price—have made it a universal favorite. Sometimes it is too bright around 6 to 8 kHz, and after countless struggles with it on horns, I finally had the sense to give up and buy a Neumann U 87. But when you hear a good C 414, the airiness and transient response are outstanding. (As with many mics, the sound can vary from one unit to the next.) These days, I rely on the C 414 almost exclusively for bass amps, and I regularly pull it out for kick drum (in omni pattern and with the 10 dB pad engaged), percussion, and congas. Truly a classic and a great buy. Price: $1,285

AKG C 3000 ***large-diaphragm, dual-pattern condenser.*** This is my long-standing nominee for best large-diaphragm condenser under $300 (street price). I have used C 3000s extensively for live-to-DAT stereo recording, as well as in the studio for overheads on bright and loud drum kits, vocals, acoustic guitar, strings, and such. Many people have bought pairs on my recommendation, and all of them have been quite satisfied. The C 414 B/ULS offers more features and superior electronics, but the C 3000 has better internal shock mounting and seems to sound less "fizzy." Other than remarks about a little extra "edge" around 3 kHz, I have heard no complaints about this microphone. Price: $438

Coles 4038 ***figure-8, ribbon.*** I am a big fan of ribbon mics, and this one is my favorite for all-around use. A trombonist friend talked me into buying one after Steve Albini used it to record his horn. This mic is unbeatable for cello, violin, acoustic bass (unless you have an RCA 44BX), low brass, and drum-room ambience. It has tons of transparent lows, a slight 200 Hz boost, and an ultrasmooth upper midrange. I've never tried it as an overhead drum mic (a common application in the Beatles' heyday), but I have received uniformly pleasing results on bass, guitar, and organ amps; all kinds of saxes; and even some vocals. The 4038 may not be bright enough for some folks, and for those raised on digital recordings and crispy condensers, it may take some getting used to. (Note: This mic is typically carried only in pro audio stores. It is also available direct from Audio Engineering Associates and Independent Audio.) Price: $1,195

Lawson L47MP ***large-diaphragm, multiple-pattern tube condenser.*** This mic is my first choice for any vocal, reed instrument, electric guitar, or organ overdub, and for any source that would benefit from the thick low mids, creamy highs, and richness that only a tube mic can deliver. The continuously variable pattern control (located on the power supply) is fun to use and gives amazingly varied results (which may delight or confound you). Furthermore, vocalists always seem to give 110 percent when you put this "golden Goliath" in front of them. I'm constantly raving about the L47MP online, as are dozens of other satisfied customers. (See "Tube Mic Têtê-à-Têtê," in the February 1998 issue of **EM** for additional pluses and minuses.) Available direct from Lawson Microphones only. Price: $1,995

Neumann U 87 ***large-diaphragm, multiple-pattern condenser.*** If I had to pick only one mic to use for the rest of my life, this would be it. The U 87 sounds good on everything, great on most things; it always delivers a big sound at any distance, with generous low end, lots of airiness, and minimal off-axis coloration. Many microphones have been modeled on the U 87's look and sound, but none offer the realistic, pleasing ambience of the real thing. Mine, which are late '70s models, are exceptionally smooth and full in the mids. My only negative comments are that the U 87 can make certain vocals sound too warm or muddy around 300 Hz, and in other applications the mic can sound a little "scratchy" due to its slight presence boost above 5 kHz. Price: $2,725

Neumann TLM 193 ***large-diaphragm, cardioid condenser.*** This mic uses the smaller diameter capsule employed in the Neumann U 89 and TLM 170 mics and offers a more neutral sound than the U 87. Although some people find it overly dark or not "zippy" enough in the highs, I find its response at a distance of two to three feet to be very flattering. I like the unique quality it imparts to potentially harsh-sounding

Legendary as a tom mic but also great on kick drum, electric guitar, horns, and vocals, the now-discontinued Sennheiser MD 421 was recently superseded by the MD 421 II (shown).

For manufacturer information, please see Contact Sheet, p. 163.

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sources, as well as the transparent lows of its transformerless circuitry. The TLM 193 has given me excellent and distinctive results in a variety of applications: on vocals, sax (including the edgy Rova Sax Quartet), trumpet, and bass amps, as well as on scratchy-sounding sources such as certain "ethnic" stringed instruments, resonator guitar, and acoustic harmonica. Price: $1,495

Oktava MC 012 ***small-diaphragm condenser with interchangeable cardioid, hypercardioid, and omnidirectional capsules and in-line 10 dB attenuation pad.*** A great "bargain" mic, the MC 012 (with the cardioid capsule) is my overhead mic of choice. This mic, excellent on acoustic guitar and percussion, has also come in very handy for soft-spoken vocalists who can't cut through on a Neumann U 87 or Lawson L47MP. The MC 012 has all the highs and snappy transient response that small diaphragms are known for, and it exhibits a full low end. Be warned, however, that prominent mids around 800 Hz (again, with the cardioid capsule) can be a problem when miking cymbals. (Note: Most stores do not carry this mic. Try Guitar Center or Oktava specialist Taylor Johnson [tjohnson@centuryinter.net], who also offers accessories, matched sets, and pretested mics.) Price: $599.99

Sennheiser MD 421 ***cardioid dynamic.*** Comparison tests in my recording classes revealed this "workhorse" mic to be a winner on electric guitar, with just the right mix of lows, cutting power, and presence for any and all musical styles. With a tube mic preamp and careful placement, it just can't be beat (except maybe by the Lawson L47MP). The MD 421 also excels on jazz and rock kick drum, is legendary as a floor-tom mic, and produces a pleasing, musical tone on horns for live-sound work. The even midrange response, classic German engineering, and unique five-position low-cut switch make this mic much more than your average dynamic. (Note: There is a new version of the now discontinued MD 421—the MD 421 II. My remarks refer to the original, since I have not yet recorded with the MD 421 II.) Price: $485 (MD 421 II)

▼

Vocalists tend to give 110 percent when confronted by the "golden Goliath."

Sennheiser MD 441 ***supercardioid dynamic.*** This is a superb dynamic mic with a sweet tonal response that rivals that of many condensers. It can be a little too bright at times, with an abundance of 5 to 6 kHz presence, but it is never harsh. The MD 441 has worked particularly well for me on "live" session vocals, guitars (acoustic and electric), and brass. (It's wonderful on tuba.) My favorite application has been with highly produced, "in-your-face" male vocals. The ample proximity effect, linked with crisp highs and a midrange dip, delivers a huge, imposing sound when worked close, and the mic is especially resistant to popping. The MD 441 also features a five-position

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low-cut switch and another switch for boosting high frequencies. Price: $895

Designed by the BBC and renowned for its smooth sound, the Coles 4038 ribbon mic is just as useful today as it was when first introduced in the mid-1950s. Here, two 4038s are positioned on a stereo bar.

Shure Beta 58A *supercardioid dynamic.* A lot of vocalists love this mic, and I used to use it a lot. It is somewhat neglected around my studio these days, because I have acquired some higher-end mics that I prefer for most applications. But I still pull it out for snare drums, guitar amps, bright hi-hat cymbals, and "live" session vocals (for example, with a loud blues band in which all the musicians are recorded simultaneously in the same room). The Beta 58A is also very good to use on trumpet when the Coles 4038 or Lawson L47MP is too sensitive or dull sounding. Basically, the Beta 58A can fill in for a Sennheiser MD 441, offering a more pronounced midrange and less "fizz" on high-presence sources. It also makes for a nice change from the standard Shure SM57 on guitar amps (as does the original Beta 58 model). Price: $332.50

HONORABLE MENTIONS

Electro-Voice RE20 *large-diaphragm, cardioid dynamic.* Another versatile and rich-sounding dynamic mic, the RE20 runs a close second to the Sennheiser MD 421 in my book. Price: $655

Fen-Tone 500-C Dual Crystal *omnidirectional crystal mic.* This made-in-Japan high-impedance mic looks like a miniature RCA 77 and is really good at sounding really bad. Run through a tube direct box, it gives the best (and cheapest) vintage sound I've ever heard: no lows, no highs—just an instant ringer for that vintage 78 rpm sound. (Available used only. Keep an eye out at pawn shops and garage sales.)

Neumann TLM 103 *large-diaphragm cardioid condenser.* This no-frills newcomer made a very positive impression during testing for **EM**'s September 1998 cardioid-condenser face-off, and some folks like it better than the Neumann U 87. It has extended low end, a hot output, and untouchably low self-noise specs. The TLM 103 is surprisingly useful on difficult sources such as vocals, horns, and strings, and for the money, it's an outstanding mic. I always grumble, however, about the flimsy plastic stand mount (I've broken two so far) and the lack of a 10 dB pad. Fortunately, Neumann has promised a new, more durable stand mount. Price: $995

Oktava MK 219 *large-diaphragm, cardioid condenser.* Another very useful and extremely affordable condenser mic, the MK 219 was best described by one reviewer as "an SM57 on steroids." The unique "vintage" frequency response is noticeably rolled off below 150 Hz and above 12 kHz, packing lots of hard-hitting midrange for vocal, snare-drum, and electric-guitar tracks. Price: $599.99

Sennheiser MD 504 *cardioid dynamic.* This is an ideal tom mic, thanks to its small size, beefy construction, low price, and superb sound. It is also good on guitars and low brass. Price: $169

Finally, I like just about any ribbon mic, including the **beyerdynamic M-130** and **M-160** ($659 each), the new **Royer R-121** ($995), the **Altec 639b** (not made since the 1950s and hard to find used), and practically all working models by RCA.

Happy miking!

Myles Boisen *is a guitarist, producer, composer, teacher, and head engineer/instructor at Guerrilla Recording and The Headless Buddha Mastering Lab in Oakland, California. He can be reached at mylesboise@aol.com.*

FM Basic Training

Algorithms and operators aren't intuitive, but they are fathomable.

By John Duesenberry

In the March 1999 issue of **EM**, I discussed the basics of modulation synthesis and explored amplitude modulation (AM) and related techniques. This article examines another form of modulation synthesis, frequency modulation (FM).

FM synthesis, in a rather limited form, is possible with voltage-controlled analog synthesizers. But the musical potential of FM didn't become fully apparent until John Chowning's pioneering work on the digital implementation of FM in the 1970s. A decade later, when Yamaha introduced the DX7 synthesizer and its many relatives, FM won mass acceptance in the music world.

The FM craze of the 1980s has abated, but one industry pundit, impressed by Yamaha's new FS1R synth ($999.95; discussed in the January 1999 "What's New" column), recently predicted an FM synth revival. If he's right, then we've picked a good time to reexamine the subject.

RON BROWN

BOOT CAMP REVISITED

For those who missed (or forgot) the earlier article, let's start with a rapid review of the general characteristics of modulation synthesis. Modulation synthesis is a waveshaping technique in which an audio-rate signal called the *modulator* controls some parameter of another audio signal, called the *carrier*. In FM, the frequency is the modulated parameter. The modulation process generates new sine-wave components, called *sidebands*, in the spectrum of the output signal. The power of the sidebands is governed by the *modulation index* (discussed shortly). The index is defined differently for AM and FM, but in both cases it is related to the amplitude of the modulator.

Sideband frequencies can be calculated by taking the sums and differences of the frequencies of carrier and modulator components. The resultant spectra fall into two broad classes. In a

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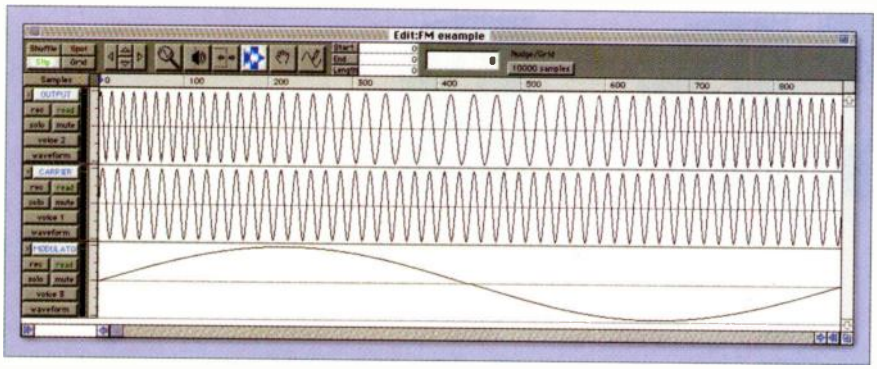

FIG. 1: FM synthesis employs at least one modulator (bottom) and a carrier (middle) waveform. The resulting waveform (top) includes new components created by the process.

harmonic spectrum, all components are members of a harmonic series, that is, they are integer multiples of some fundamental frequency. In other words, their frequencies are 2, 3, 4, and so on, times the frequency of the fundamental. In an *inharmonic* spectrum, some or all components do not fit into a harmonic series.

The ratio of the carrier and modulator signals' frequencies, which can be represented as F_c:F_m, determines whether a modulation spectrum will be harmonic or inharmonic. Here we repeat Rule 1 for fledgling modulation synthesists:

Rule 1. If F_c:F_m is a ratio of simple integers, the modulation spectrum will be harmonic. Otherwise, the spectrum will be inharmonic.

Now let's relate these generalities to the specifics of FM.

SIMPLE FM SPECTRA

One reason for FM's popularity is that interesting spectra can be synthesized with limited resources. In FM, a sine carrier and modulator generate a theoretically infinite number of sidebands. By varying one simple parameter, the modulation index, we can create complex variations in the spectrum. Sine-wave FM with dynamic index control (that is, an index that changes over time) is the basis of most commercial FM synthesizers.

Figure 1 shows the output waveform that is created from a typical sine carrier and modulator. The waveshaping effect, a kind of "bending" of the sine waveform as its instantaneous frequency changes, is apparent. Though this may be visually interesting, the spectrum that results is even more so.

The resulting spectrum of an FM process is easy to predict. Given a sine carrier of frequency F_c and a sine modulator of frequency F_m, the FM spectrum will consist of the following components:

Upper sideband frequencies, which are the sum of F_c and every integer multiple of F_m ($F_c + F_m$, $F_c + 2F_m$, $F_c + 3F_m$, and so on).

Lower sideband frequencies, which are the difference of *Fc* and every integer multiple of F_m ($F_c - F_m$, $F_c - 2F_m$, $F_c - 3F_m$, and so on).

The original carrier frequency, F_c.

Let's consider the spectrum resulting from FM with a 500 Hz carrier and a 400 Hz modulator. The ratio F_c:F_m reduces to 5:4, so according to Rule 1, this will be a harmonic spectrum. **Figure 2a** represents the first three sideband pairs around the carrier. Notice that some lower sidebands have negative frequencies. This will occur whenever F_m or one of its multiples is greater than F_c. A signal with frequency $-F$ is simply inverted (180 degrees out of phase) with respect to a frequency F. In this particular example, the negative frequencies wouldn't affect the sound.

In **Figure 2b,** negative components are represented as having negative amplitudes (downward lines). In fact, they are positive, but 180 degrees out of phase, as noted earlier. This makes it easy to see that the spectrum is harmonic, as predicted. It consists of a

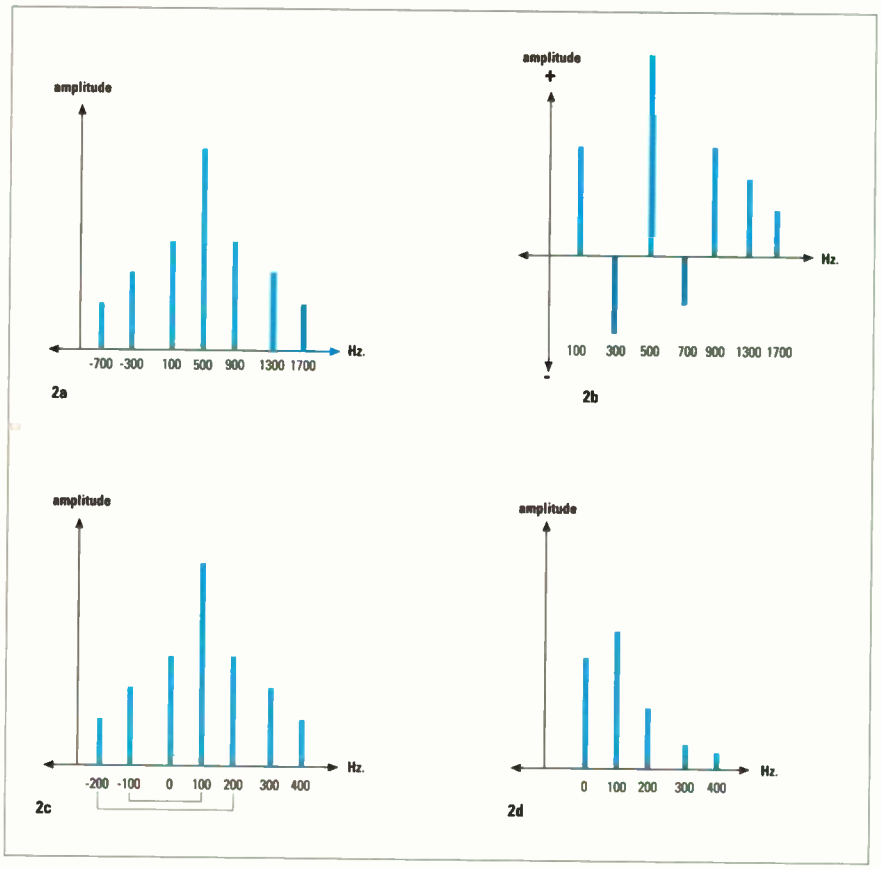

FIG. 2: Different FM spectra result from a carrier waveform (F_c) being modulated by a modulator (F_m). The resulting waveforms occur in pairs called "sidebands." When F_c = 500 Hz and F_m = 400 Hz, sideband pairs appear around the carrier (2a); the "negative" components are actually positive but 180 degrees out of phase (2b). When F_c and F_m = 100 Hz, you get the spectrum in 2c. Figure 2d shows the spectrum that results when every positive component in the spectrum is attenuated.

For manufacturer information, please see Contact Sheet, p. 163.

100 Hz fundamental, with odd-numbered harmonics. Don't confuse the carrier component (500 Hz) with the fundamental (a 100 Hz sideband).

Figure 2c shows how negative frequencies affect the sound. This is another harmonic spectrum, because F_c and F_m are both 100 Hz (F_c:F_m = 1:1). Although the sideband series around the carrier extends infinitely, only the first three sideband pairs are shown. The first lower sideband has a frequency of 0, which is an inaudible DC component. As the brackets show, the 100 Hz carrier component is matched with a sideband of its inverted frequency (–100 Hz), as is the 200 Hz component.

The brackets indicate a pattern that holds throughout this spectrum: for each positive, nonzero component, there is a corresponding negative component of unequal amplitude. The summation of the corresponding positive/negative components produces a partial cancellation, or attenuation, of every positive component in the spectrum. **Figure 2d** illustrates this result. The audible spectrum has a fundamental of 100 Hz, with all harmonics present.

Here's a quick quiz: compute the first three FM sideband pairs where F_c = 500 Hz and F_m = 202.61. Is this spectrum harmonic or inharmonic? (Answer: The lower sidebands will appear at 297.39, 94.78, and -107.83. The upper sidebands are 702.61, 905.22, and 1107.83. The spectrum is inharmonic.)

THE MODULATION INDEX

Although the number of FM sidebands is infinite, there is a finite number of *significant* sidebands. A significant sideband pair is one that has more than 1/100 of the amplitude of the carrier. The FM modulation index, or I, governs both the number of significant sidebands and their relative amplitudes. Before I explain the effect of the index, I need to define a couple of additional terms.

The instantaneous frequency of the modulated carrier deviates above and below F_c in proportion to the amplitude of the modulator. In *linear* FM, the positive and negative frequency excursions, measured in hertz, are equal. I will assume linear FM in this discussion, because digital FM implementations are normally based on linear FM.

The maximum change from F_c is the *maximum frequency deviation,* or D. You can think of D as the "depth" or "amount" of modulation. The FM modulation index I is defined as the ratio of frequency deviation to modulator frequency: $I = D/F_m$.

As I increases, the number of significant sidebands increases, while the carrier component is weakened. The sound of FM with a slowly increasing index is distinctive, resembling an elaborate crossfade between the sine-wave carrier and a number of partials above and below it. As I decreases, of course, the spectrum evolves in the opposite direction, and the sidebands disappear.

This overall change in sideband power doesn't mean that the amplitudes of the individual sidebands all change by the same amount. In fact, as I changes, the amplitude of each sideband pair evolves in a different pattern. As some sidebands gain amplitude, others lose amplitude and disappear. In addition, there may be

cancellation effects caused by phase-inverted sidebands. This accounts for the complex "churning" quality of a dynamically changing FM spectrum.

The amplitude of a particular FM sideband for a known value of *I* is given by a mathematical formula called a *Bessel function*. FM synthesists don't spend their lives computing Bessel functions, of course. But if possible, take a glance at the Bessel function plots in *Computer Music*, 2nd edition (Dodge and Jerse; Schirmer, 1997). They will help you picture how the modulation index affects the sidebands.

What does all this mean to the musician? A dynamically changing index generates a dynamic spectrum that holds the interest of the ear.

OPERATORS AND ALGORITHMS

To obtain a dynamic index, you need only control the modulator amplitude with an envelope that is scaled to the modulator frequency. Package this function into a little bundle called an *operator*, and you have the basic building block of Yamaha-style FM synthesis.

An operator can function as either carrier or modulator. **Figure 3a** illustrates the simplest configuration of two operators: the output of operator 2 is routed to the frequency-control input of operator 1. The envelope generator within operator 1 controls the final output amplitude of the patch.

Yamaha FM synthesizers feature various fixed configurations of operators. These configurations are called "algorithms" (which sounds more impressive than "patches"). **Figure 3b** shows our two operators as part of a Yamaha-style algorithm. In this algorithm, operator 2 modulates operator 1; operator 4 modulates operator 3; and operator 6 modulates itself (via feedback) and operator 5. The outputs of the carrier operators 1, 3, and 5 are summed.

PORTRAIT OF A SERIAL MODULATOR

An understanding of simple 2-operator sine-wave FM is essential if we are to understand more complex FM synthesizers, which usually employ at least four operators patched in many different ways. The DX7 had six operators and 32 algorithms; the new Yamaha FS1R has eight operators and 88 algorithms. Not wishing to seem too Yamaha-centric, I should point out that there are other approaches. The original Synclavier, for example, used a sine modulator and a complex carrier that was created using additive synthesis. Software synthesis systems place few restrictions on the number of FM oscillators, their waveforms, or their interconnections. Let's consider some of the possibilities.

In *multiple-carrier* FM, a single sine modulator controls more than one carrier. The result is the sum of the three modulated carriers; their spectra are superimposed. With harmonically tuned carriers, peaks, or *formants*, in the spectrum can be produced. Composers such as John Chowning and Dexter Morrill have simulated vocal and brass timbres in this way. Note that modulating a complex carrier, such as a sawtooth wave, is an instance of multiple-carrier FM. In this case, the carrier can be analyzed as some number of sine components, all modulated by the same signal.

If that's not complicated enough, consider *multiple-modulator* FM, in which a single sine carrier is controlled by several modulators. The modulators may be connected in parallel or in series. In **Figure 3c**, operators 2 and 3 are patched in parallel to modulate operator 1. With two modulators, each sideband generated by one acts as a carrier that is in turn modulated by the other. This can generate a huge number of sidebands, but you can keep them under control by simply using low modulation indices.

The use of complex modulators, such as sampled or additively synthesized signals, can also be regarded as examples of parallel, multiple-modulator FM. In such cases, just think of the complex modulator as the sum of a large number of sine modulators.

Modulators can also be connected in series, as in **Figure 3d**. In this "cascaded" or "chained" configuration, operator 3 modulates operator 2, producing a complex signal that in turn modulates operator 1. In practice, serial-modulator FM spectra are very similar to parallel-modulator spectra.

OTHER FM OPTIONS

Modulating the frequency of a periodic carrier with a noise signal generates random sidebands above and below the carrier. This is an excellent way to obtain a "pitched noise" effect that is similar to noise filtered through a narrow bandpass filter.

Feedback FM, where the output of an oscillator is fed back into its own frequency-control input, is a technique patented by Yamaha. In feedback FM, the number and amplitude of the sidebands tend to increase in a more linear relationship to the modulation index. This spectral evolution is closer than simple FM to the natural evolution of acoustic instruments' spectra.

FM SYNTHESIS TOOLS

Mention FM synthesis and Yamaha comes to mind. If you are going to get involved with FM, you'll certainly want to consider Yamaha products. Yamaha claims that their latest FM box, the FS1R, combines FM with "formant-shaping synthesis." Yamaha's current lineup also includes the EX5 (reviewed in the March 1999 issue of **EM**) and EX7, which feature a form of analog modeling that supports FM.

You can find older Yamaha FM synthesizers second-hand at bargain prices; the models are too numerous to list here. I personally prefer the 6-operator models—I'll never give up my ten-year-old TX816—but 4-operator units are cheaper. The Yamaha SY-series products allow modulation of FM operators by AWM2 samples.

If you want to hook up your own operators and algorithms, software synthesis is the way to go. You can find links to MIT's *Csound* (cross-platform) and James McCartney's *SuperCollider* (Mac) at Tom Erbe's Mac software Web site (shoko.calarts.edu/~tre/CompMusMac). GUI-based synthesis/DSP programs include Jim Bumgardner's *Syd* (Macintosh; also available from Erbe's site), Seer Systems' *Reality* (Windows), Synoptic's *Virtual Waves*, Digidesign's *Turbosynth SC* (Macintosh), Cycling 74's *MSP* (Mac), and Symbolic Sound's Kyma System (which uses dedicated hardware and Mac or Windows software). All of these support FM synthesis in some form.

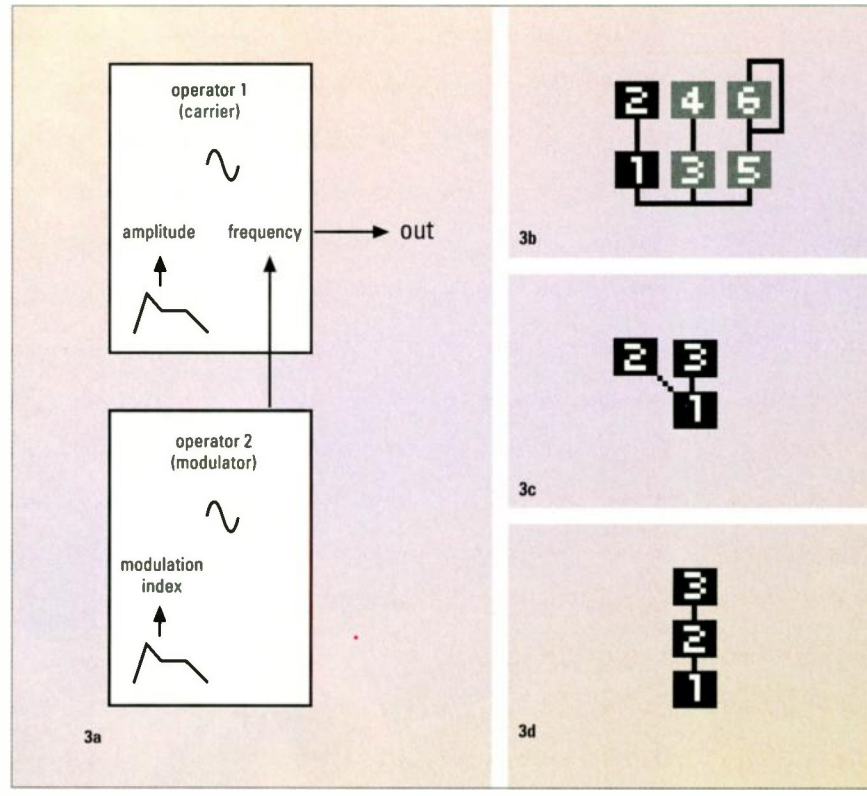

FIG. 3: These four FM synthesis configurations are typically called "algorithms": 3a is a conventional signal-flow diagram with two operators; 3b is Yamaha-style "algorithm" notation with feedback FM, in which the output of oscillator 6 is fed back into its own frequency-control input; 3c shows parallel multiple-modulator FM, in which operators 2 and 3 modulate operator 1; and 3d plots series multiple-modulator FM, in which operator 3 modulates operator 2, which modulates operator 1.

Feedback FM with a very high modulation index can yield extremely rich spectra, sometimes resembling high-frequency noise.

Finally, a note to all you analog fans. The classic linear FM sound, as implemented in most digital FM synthesizers, is all but impossible to obtain on most analog synthesizers. Analog oscillators are typically designed to have a 1-volt-per-octave response to keyboard controllers. This is usually accomplished by putting an *exponential converter* on the frequency-control input. Because of the converter, an incoming sine modulator will drive the carrier frequency asymmetrically. The positive frequency deviation will be greater than the negative deviation. This raises the perceived pitch of the modulated signal. Consequently, changing the modulation index has a pitch-bending effect, and it is difficult to get an exponential FM patch to sound in tune across a wide pitch range. Exponential conversion also distorts the modulator signal, turning a sinusoidal modulator into a complex waveform.

This doesn't mean that exponential analog FM sounds rotten, just that it sounds different from the FM effects we're used to. Analog oscillators with a linear FM response have been built, but they've never been plentiful. The Moog modular 901B VCO had a linear input, as did VCOs made by Serge Modular and Gentle Electric. With other gear, it may be possible to modify the hardware and bypass the exponential converter.

VALEDICTION

As I wrote this article, I enjoyed revisiting my Yamaha TX816 and taking another look inside its algorithms. I hope you'll want to get some hands-on experience with this versatile technique, too (see the sidebar "FM Synthesis Tools"). Whether FM synthesis is in fashion or not, it has great potential for producing interesting sounds, and every synthesist should be acquainted with it.

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

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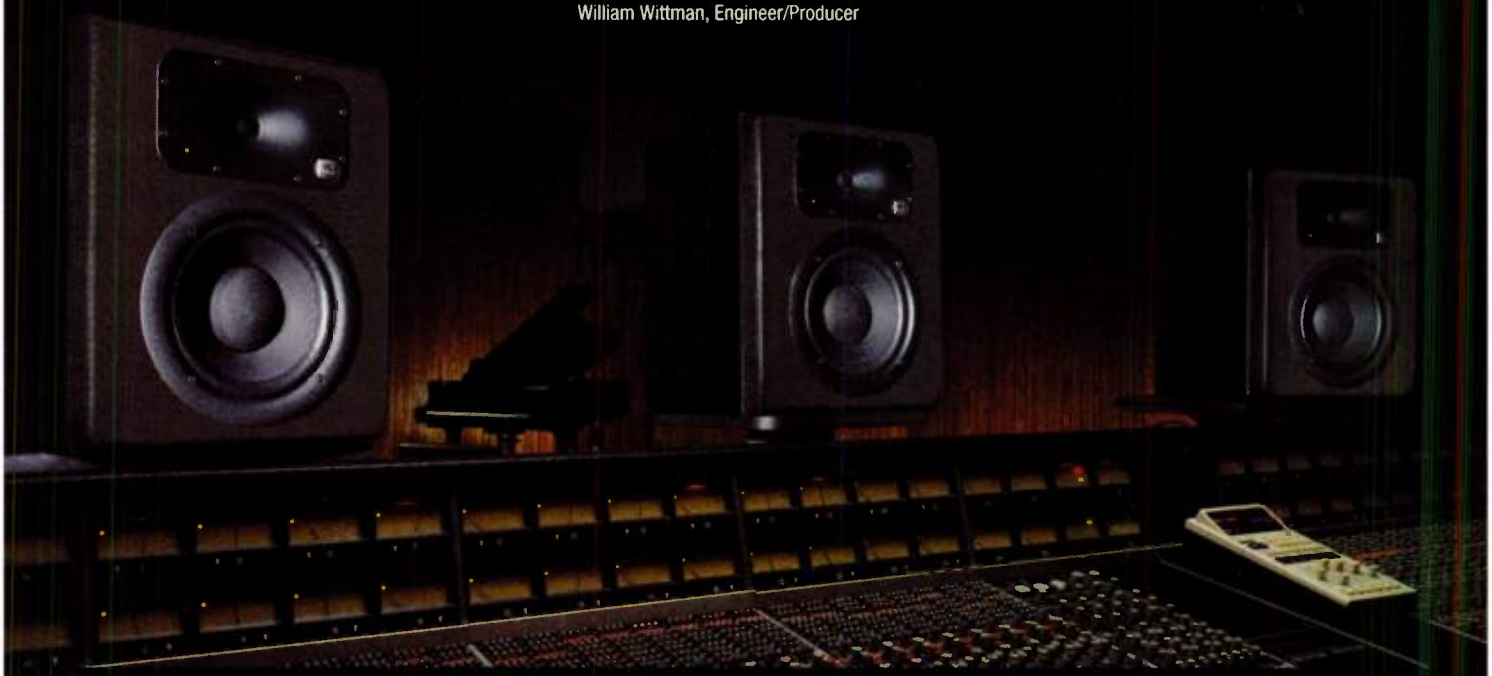
William Wittman, Engineer/Producer



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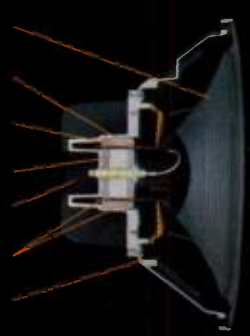

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Taxi Undercover

This A&R vehicle doesn't just take you for a ride.

By Joanna Cazden

Seven hundred and fifty songwriters and performing artists crowd a hotel conference room near L.A. International Airport. Randy Bachman, who penned such hits as "Taking Care of Business" for Bachman Turner Overdrive and "These Eyes" for the Guess Who, is on stage, relating the tale of how he unsuccessfully pitched a certain song to a country-music star on a dozen different occasions. A couple of years later, the artist was looking for new material, and Bachman submitted the demo once more. This time, the star said, "It's terrific! You rascal, why haven't you played me this one before?"

The crowd erupts with groans and laughter about the fickleness of the music industry. In the back of the hall, conference host Michael Laskow grins with pride. As the founder and president of a unique organization called Taxi ("the independent A&R vehicle"), he can tell that his passengers are getting the message. All songwriters—even those such as Bachman who have superlative credits—face the same difficulties in getting their material heard. But Taxi helps by acting as a source filter, or as Laskow puts it, "a casting agency for music."

JACK DESROCHER

UNDER THE HOOD

There are no prerequisites for joining Taxi. Membership costs $299 for the first year, and annual renewal is $199. Every two weeks, members receive a list of song and artist requests from labels and producers (see the sidebar "Sample Song Listings"). If you have material that you think fits a listing, you submit it with a simple form.

Taxi's music screeners listen to everything that comes in. Cuts that are judged suitable for each listing are forwarded with the prestige of "solicited material." The rest are returned to their owners with a critique (see **Fig. 1**). You can send in any number of tapes, but a $5 submission fee per tune encourages members to be selective. "I didn't want

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to charge anything for sending in songs," explains Laskow. "But John Braheny of the Los Angeles Songwriters Showcase advised me to set a nominal fee just so people would stop and think about what they send." For $10, you can send in material without reference to a listing to receive feedback on your songwriting, your production, or even your CD packaging or press kit.

STARTING THE ENGINE

Laskow founded Taxi after many years as an engineer, producer, and studio manager. Although he worked with musicians such as Eric Clapton and Crosby, Stills, Nash and Young, he got a deep personal satisfaction from helping unsigned bands. He resolved to find a way to help open industry doors to unknown but deserving talent.

Songwriter advocate Braheny was skeptical when Laskow approached him for startup advice. His experience with scam artists and naïveté in the music business had made him wary of such enterprises. But Laskow seemed both knowledgeable and determined to keep his business clean in image and in reality. Braheny became his first screener and remains a key advisor.

Taxi began its outreach in 1992. The organization's membership has doubled every year since then and is now in the thousands. The core staff includes seven full-time employees and several dozen screeners. The number of active listings at any one time has grown from about 20 to near 60, due in part to Taxi's exceptional screening process.

As Braheny explains, "Over the years, there have been lots of tip sheets [insider listings] for songwriters. But the A&R [Artist and Repertoire] people would get deluged with hundreds of poor-quality tapes, so they'd stop soliciting work, and the tip sheets would fold. The industry has come to trust Taxi to forward only the cream of the songwriting crop."

Jai Josefs, a songwriter and producer and author of *Writing Music for Hit Songs* (Simon & Schuster), has screened songs at Taxi for two years. He, too, speaks enthusiastically about the company.

"When I started with Taxi, I was already teaching songwriting seminars around the country, so I knew how to give helpful feedback. The screeners tend to be friendly and approachable folks. But many worked in A&R situations where they judged songs or bands on a yes or no basis only: a great hook right away, or forget it. Taxi gets its screeners started with John Braheny's book *The Craft and Business of Songwriting* [Writer's Digest Books] and gives them suggestions on how to write appropriate critiques for the songs they reject."

SCOTT WILKINSON

Attendees at Taxi's Road Rally vote on the quality of the song they've just heard. Randy Bachman is the long-haired fellow seated in the left foreground.

Josefs adds, "Although each of us usually screens in a few different genres, we're never asked to critique something totally outside of our expertise." And Braheny emphasizes that the screeners *want* to be blown away: "We send back a lot of what we hear, but we definitely come to work hoping to find exciting material."

Every couple of months, the screeners are treated to a free dinner and training seminar, such as an update on the ever-changing flavors of the marketplace or a round-table discussion about songs that fall between commercial genres. According to Laskow, "We figure that if we treat our screeners right, they'll be more likely to treat the members right."

TEST-DRIVE

Now, I've been writing songs for more than 20 years, and a few bands have covered my tunes. I've attended classes on commercial songwriting technique and released several self-produced albums. I was curious to see how Taxi's screeners would respond to my work.

So far, I've sent in about ten songs for four or five listings. I've had no big breaks yet, but I do feel that I've been heard fairly. I've sent the same song for several listings and received different comments on why it wasn't quite right. And although the critiques occasionally left me humbled, I never felt trashed or personally attacked.

Shortly after I joined Taxi, I hand-delivered a demo to its office in Woodland Hills, California, to meet a submission deadline and perhaps get a glimpse of the inside operation. I fully expected to give my cassette to a

Michael Laskow steers Taxi through the A&R minefield, helping members improve their craft and producers discover new talent.

For manufacturer information, please see Contact Sheet, p. 163.

harried secretary and be back on the freeway in minutes.

Instead, I was met at the office door by Michael Lederer, Taxi's cofounder and chief financial officer. This laid-back fellow with a wide grin invited me in to look around, and he took plenty of time to answer my questions, without knowing I was researching an article.

At one end of the comfortable but no-frills office sat five guys at little tables, each with a boom box and headphones, listening and writing intently. At the other end of the office were rows and stacks of white mail bins with labels bearing the names of Geffen, Arista, MCA, and dozens of lesser-known labels and production companies.

Lederer explained that the number of tapes that get forwarded to a label varies from 2 to 40 percent of those received for each listing, depending on how picky Taxi has been asked to be; the average is 11 percent. A bit less than half of the forwarded material actually results in a deal.

Laskow added that about 40 percent of Taxi members will get at least one demo forwarded per membership year.

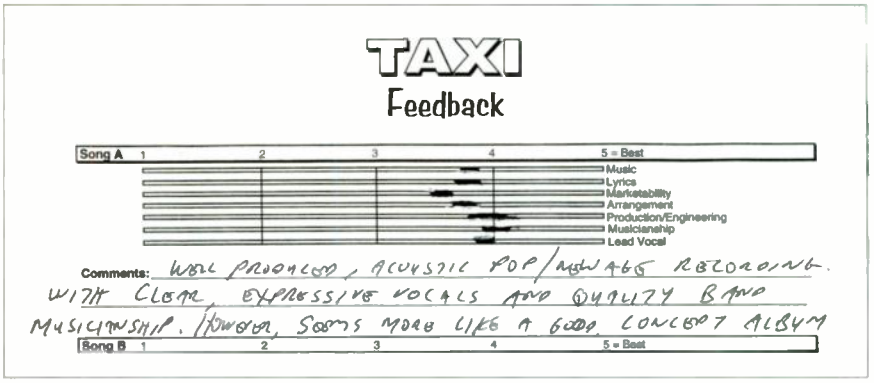

FIG. 1: This critique indicates that the music, lyrics, marketability, arrangement, production/engineering, musicianship, and lead vocal of the submitted recording are all pretty good (rated at around 4 out of 5). The handwritten comments give more detailed feedback on the song's style.

More than 450 deals have been struck since 1992, ranging from single-song contracts for albums or films to staff-writer jobs to major-label deals for bands. The highest success rate is in television and film placements, which have earned Taxi writers thousands of dollars in royalties.

The company makes no guarantees that members will get their material signed, only that it will be heard. According to Lederer, "Overall, the quality of submissions keeps going up. More skilled writers are recognizing that we're for real and are using our service. But we're just as proud that members who stay with Taxi tend to become better writers as a result of our feedback."

About half of the members renew each year, which attests a high satisfaction rating. I plan to be among the renewals this year.

PARTS DEPARTMENT

Taxi offers plenty of opinions about song production. When asked for tips at the Road Rally conference, Bachman urged composers to invest in their own gear. "That way," he said, "you can remix your tunes as fashions change and dub off copies in a flash."

Laskow's suggested gear for recording demos includes an ADAT, a Shure SM57 microphone, one or two condenser mics in the $300 to $400 range, three channels of compression, a basic multi-effects box, and a 16- or 24-input mixer. For most genres, he recommends having a strong bottom end with plenty of bass and kick drum, and the lead vocal up front.

COURTESY TAXI

FIG. 2: Taxi's Web site provides lots of useful information.

Because Josefs screens songs in several genres, he tunes his recommendations to fit. "Country or singer/songwriter submissions do just fine with, say, a Roland VS-880 or VS-1680, some decent vocal mics, and live backup players. Country music especially needs that live sound. For dance music, pop, R&B, and so on, you can do more sequencing, perhaps with Emagic's *Logic Audio* program and a Roland JV-2080 sound module."

Josefs also points out that sending in a tape with a shaky or off-pitch vocal just doesn't cut it. "If you have only $300 to invest in your demo," he says bluntly, "spend $200 on a strong vocalist."

Taxi screeners and Road Rally panelists emphasize that a song demo has different requirements than a package pitching an artist. If the material will be rerecorded by another performer, production can be simple. The melody, lyric, and groove should be prominent, without a long intro or overly dramatic instrumentation. However,

SAMPLE SONG LISTINGS

The following listings were sent out to members of Taxi during November 1998:

Major music publisher looking for outstanding, consistent modern rock songwriters with songs that would be recordable by today's charting modern rock/AAA artists. We've been asked to be extremely selective for this listing.

Music supervisor for major daytime soap opera looking for master-quality, sexy R&B/pop songs à la Anita Baker, Whitney Houston, etc., to be used in love scenes. Your version will be used—they won't rerecord.

U.K. production music library looking for instrumental/underscore composers who can produce broadcast-ready work for use in television. Please submit a montage of your work totaling no more than five minutes. They want people who are capable of writing short cues ranging from 10 seconds to 1:45 at the longest.

Major Christian publishing company looking for new and innovative, singable praise and worship songs for various projects. They want an acoustic/AAA sound, all tempos. They prefer songs that are currently being used in contemporary worship services and are accessible to most contemporary church congregations.

Senior director of A&R at major Nashville label looking for harmony-oriented mainstream country (not too traditional, not too pop) for a major country band à la Alabama. Family oriented, workin' man, or patriotic themes would be appropriate. Definitely no drinkin' or cheatin' lyrics for this listing.

if master quality is requested, the tape or CD must sound as complete and professional as possible.

OFF-ROAD, ONLINE

Taxi recognizes that musicians are increasingly Web savvy. Its Web site offers monthly chats with industry professionals, an international contact and referral service, online versions of Taxi's listings and monthly newsletter, background information on the company and its staff, and a membership form (see **Fig. 2**).

By this spring, Taxi plans to give each of its members a free Web site, including graphics, a description of their music, and streaming audio of up to three songs. A $70 premium service will offer higher-bandwidth streaming audio, a personal chat room, and, eventually, direct-sales capability.

Taxi has also experimented with music publishing and tried its hand at managing bands. But the staff has decided to stay focused on what the company does best: helping songwriters be heard, teaching them how to increase their chances of coming out with a breakthrough hit, and assisting labels and producers in finding new writing talent.

CONSUMER REPORTS

Among my friends are musicians who are generally critical of the music business, which they see as money driven and amoral. I was curious about how they would react to Taxi.

When I mentioned the company to some of them, I might as well have suggested visiting a den of wild dogs. "Those A&R boys wouldn't know good music if it got in bed with them. They're just playing the promotion game and looking to score." This was one of the gentler comments. Others complained about high membership fees.

When I repeated these comments to Laskow, he was sympathetic. "People who are that angry have probably dealt with the major labels directly and have gotten burned. This business certainly has its share of jerks. But in the long run, a good song is a good song, and Taxi is trying to get the best songs heard. Besides, membership costs less than people spend on a piece of gear or one good songwriting class."

Laskow went on, "We regularly get Internet spam attacking us. We check our database against these messages, and 99 percent of the criticism comes from people who've never been members. Those who take the time to get the real story are very loyal to us."

At the Road Rally, I chatted with many folks who had come long distances to attend. One said, "I'm from North Carolina, and there's no music school or songwriter network anywhere nearby. The information and the camaraderie here are totally worth it." Most seemed to accept that Taxi's mission is not to transform the music business, but rather to help talented artists and writers succeed in the business as it is.

SHOCK ABSORBERS

If Taxi has a hidden agenda, it's to make the independent A&R process as fair and humane as possible. According to Laskow, "My grandfather taught me that an honest businessman gives customers more than they ask for. As soon as we hear about any kind of problem, we clean it up right away."

Josefs agrees. "Taxi is a straight-ahead company. I've never heard any of the staff tell one story within the office and then put on a different face for members or the public. They say what they do, and they do what they say. The only unhappy members I know of are those who can't be objective about the quality of their own work, who have unrealistic expectations and get mad at the critiques instead of learning from them."

By the way, Randy Bachman has been a member for nearly four years. After hundreds of submissions to Taxi listings, he has several songs on hold (optioned for serious consideration) with major Nashville producers. Just like everyone else, he is told to get to the hook faster, but he doesn't give up.

At the end of the Road Rally, I was left with the impression that Taxi is a bit like the lotteries that states use to fund public schools: whether or not you win the grand prize, you've invested in education. Taxi offers prizes both large and small, and the education is yours to keep. For anyone serious about the songwriting business, Taxi provides a reality check and a cheering squad. I'd say it's worth the ride.

Joanna Cazden *is a singer/songwriter in Southern California. Her first submission to Taxi didn't get to the hook fast enough.*

REVIEWS

ROLAND

VS-1680

A truly professional portable digital studio.

By Rob Shrock

Admittedly, my problem was enviable by most people's standards. I was scheduled to tour with Elvis Costello and Burt Bacharach for the entire month of October 1998. But I had already been hired to produce an album that had to be delivered by October 31. What to do?

My only choice was to assemble a recording rig that could be set up easily in hotel rooms and concert halls. I wanted to keep the travel gear to an absolute minimum—excess-baggage charges can kill you—without compromising the audio quality of the recording.

The obvious solution appeared to be a complete Roland VS-1680 portable digital studio. Roland's VS-880 (reviewed in the September 1996 EM) has garnered a loyal following, thanks to its sound quality and comprehensive features. The latest and greatest in the VS series—the VS-1680—is designed to provide an even more powerful and complete recording environment.

The VS-1680 combines a 16-track hard-disk recorder; an automated, 26-channel mixer (10 channels dedicated to the 8 analog and 2 digital inputs, and 16 channels for the disk-based tracks); and slots for two optional stereo effects-processing cards. The 320 × 240–pixel graphic LCD is a vast improvement over

Roland's VS-1680 portable digital studio can record on eight tracks simultaneously and output four tracks in the digital domain. Its large LCD not only shows the parameter values, playlist, status information, and location; it also lets you zoom in to view your tracks at the waveform level.

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FIG. 1: The VS-1680's eight RCA outputs (center) can serve as individual channel outputs or can be grouped into pairs to function as the master, aux, and monitor outs. Each pair is assigned separately. The coax and optical digital S/PDIF outputs (lower right) can carry different signals simultaneously, but the digital inputs are an either/or proposition.

the VS-880's small window, allowing you to view waveforms easily. An internal 3.1 GB hard drive provides data storage.

If you buy the VS8F-2 effects cards ($395 each) and use an optional CD-RW SCSI drive ($750) for data backup or creating audio CDs, you have everything you need except a pair of powered monitors, a microphone, and, of course, a source to record. Equipped with this promising new tool and a few other pieces from my studio, I hit the road, hoping that the VS-1680 wouldn't let me down.

TRACKS AND V-TRACKS

Channels 1 through 8 are mono; channels 9 through 16 are configured as four stereo pairs. The stereo pairs can be split into independent mono channels that share the same fader. The VS-1680 can play back 16 tracks simultaneously, and each track (or stereo pair) has 16 V-tracks (virtual tracks). You can play back one V-track for each regular mixer track; think of the recorder as 16 tracks wide and 16 tracks deep.

Using up to 256 V-tracks, you can do multiple takes of a performance, explore different arrangements, submix sections, and much more. When you get an idea, simply grab another V-track, record a new take, and sort it all out later.

RECORDING MODES

The VS-1680 employs a new version of Roland's R-DAC data compression to maximize recording time. It has six recording modes, which use various amounts of compression and offer proportional amounts of recording time (see the table "Recording Time"). Each mode can operate at 48 kHz, 44.1 kHz, or 32 kHz. You cannot mix recording modes within a song.

The new MTP (Multi-Track Pro) mode is best for high-caliber, 16-track recording. It uses the least lossy compression algorithm of the bunch and can record up to 24-bit files. In this mode the VS-1680 accepts 24- or 16-bit audio input from the S/PDIF ports or 20-bit digital audio via the analog inputs and A/D converters.

Four 16-bit, 16-track recording modes are available. Unlike MTP mode, these use a data format that is compatible with VS-880 files. MT1 (Multi-Track 1) mode employs a soft compression algorithm. MT2, LIV1 (Live 1), and LIV2 each have progressively more data compression, and therefore progressively lower audio quality. If you just can't live with any kind of compression, you can record in 16-bit MAS (Mastering) mode, which does not use compression but reduces the number of available tracks to eight.

The compressed modes result in compression ratios that vary from approximately 2:1 to 4:1, which is impressive, considering how good the playback sounds. Roland declined to provide any specific details or exact numbers regarding its R-DAC compression scheme, but the three best (lowest compression) modes sound excellent. The compression has obviously been improved over what was used in previous VS-series workstations.

In fact, the VS-1680's compression scheme is so good that I couldn't really hear any degradation in the sonic quality of the source material when recording in MTP, MT1, and MAS modes. Maybe the top end wasn't quite as in airy as in the original source, but we're talking about a hairsplitting difference. As you move into higher compression modes, you do begin to hear added noise and graininess, making these modes more appropriate for archiving noncrucial low-fidelity audio, documenting lectures, or recording rehearsals.

Depending on what you are doing, it might be a good idea to experiment with using one of the top-quality compression modes at a 32 kHz sample rate. Although you lose the higher frequencies and upper harmonics, this setting still sounds quite good for certain applications, and it gives you much

VS-1680 Specifications

A/D Converters	20-bit, 64x oversampling
D/A Converters	20-bit, 128x oversampling
Internal Processing	24-bit (mixer section)
Sample Rates	48, 44.1, 32 kHz (22 kHz–50.48 kHz range w/ Vari-Pitch)
Frequency Response	20 Hz–22 kHz (±0.2 dB @ 48 kHz); 20 Hz–20 kHz (±0.2 dB @ 44.1 kHz); 20 Hz–14 kHz (±0.2 dB @ 32 kHz)
Total Harmonic Distortion	0.005% (MTP mode); 0.003% (MAS mode)
Residual Noise Level	-82 dBu
Nominal Input Level	-50 to +4 dBu; +26 dBu maximum (balanced); +20 dBu maximum (unbalanced)
Input Impedance	30 kΩ (inputs 1–8); 500 kΩ (guitar input)
Output Impedance	1 kΩ
Maximum Storage Capacity	128 GB (eight 2 GB partitions per disk drive x 8 drives)
Display	320 x 240–pixel, backlit graphic LCD
Dimensions	21.8" (W) x 4.3" (H) x 13.25" (D)
Weight	13 lbs., 15 oz. (without hard drive)
Phantom Power	48V

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*Manufacturer's Suggested Retail Price (US)

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more recording time per megabyte of disk storage. This could be a good choice when you're recording music or dialog for computer games and multimedia; these applications typically require a lot of recording time and don't need extended top end.

THE STORAGE STORY

As I noted earlier, the VS-1680 comes with a 3.1 GB internal hard drive. You can use a larger drive, but it must be partitioned. (The largest partition the VS-1680 supports is 2 GB.) With a maximum of eight drives (one internal, seven through SCSI), each with up to eight partitions, you could have as much as 128 GB of storage available at a time.

Practically everything—audio files, mixer settings, routing, automation information, and Song parameters—can be stored in a data-only backup session. You can back up the audio to a DAT machine through the S/PDIF outputs, or you can copy data files to a SCSI device, such as a CD-RW unit. (By the way, the VS-1680 can add SCMS copy protection to DAT copies, if you wish.)

You can record two Red Book audio tracks on CD-R to create submasters that will play in a consumer CD player. With the VS-1680's current operating system, you have to mix down to 2 of your 16 tracks. However, in a future software update, Roland plans to add an internal stereo pair (tracks 17 and 18) for storing mixes.

GETTING IN AND OUT

While the VS-880 has only four audio inputs, which limits its usefulness for recording live ensembles, the VS-1680 offers eight simultaneously available channel inputs (see **Fig. 1**). Channels 1 and 2 have balanced XLR inputs with phantom power; unfortunately, they don't have ¼-inch line-level inputs. Channels 3 through 8 have ¼-inch TRS mic/line inputs without phantom power. A separate, high-impedance guitar input (which overrides the regular channel 8 input) is provided with the VS-1680, effectively giving you a built-in direct box.

The eight RCA outputs can serve as individual channel outputs or can be grouped into pairs to serve as the master L/R, aux A L/R, aux B L/R, and monitor L/R outputs. (The aux sends can be pre- or postfader.) Each pair is assigned independently, so it's possible, for example, to assign outputs 1 and 2 as master outputs and outputs 3 and 4 as direct outs.

Stereo S/PDIF digital inputs and outputs are provided in both coax and optical formats. You can't use the coax and optical inputs simultaneously, but you can use both sets of digital outputs at the same time. Even better, the two digital output pairs can be assigned independently. This could come in handy, for example, when creating multichannel mixes of separate elements for TV or video.

The mic preamps are excellent, and I used them quite a bit on instruments. In fact, one of the best acoustic guitar sounds I got was accomplished using the internal mic preamps (with no compression). I still prefer a custom Audio Upgrades outboard mic preamp for vocals, but if you have to record tracks with a minimum of fuss, the VS-1680 preamps won't let you down.

THE QUICKEST ROUTE

The VS-1680's automated mixer makes it easy to create complex mixes and sort through all of your V-tracks, which have a tendency to multiply like political scandals. The internal routing is extremely flexible. I especially like the ability to assign tracks using dedicated buttons, so you don't have to wade through menus on the LCD.

For example, to assign input 1 to track 4, you simply hold down the Input 1 button while pressing the Track 4 button; the Track button's LED will light green. Any time you press the button for input 1, the LEDs for all tracks fed by that input will light up. The process works the same in reverse: when track buttons are held down, the associated input LEDs light up. This makes very quick work of creating and verifying assignments, thus reducing routing mistakes.

The EZ Routing feature lets you store routing assignments for all inputs and outputs on the mixer, including the input channels' track assignments and settings for effects sends, aux sends, V-tracks, effects inserts, channel level and pan, EQ mode (2-band or 3-band), and stereo link status (see **Fig. 2**). EZ Routing has many uses. For example, if you have an elaborate input map for a drum kit, you can store it for later recall.

For manufacturer information, please see Contact Sheet, p. 163.

VS-1680 Features

Tracks (record/playback/virtual)	8 (simultaneous)/16/256
Mixer Channels	26 (8 mono tracks; 4 stereo/dual mono tracks; 8 analog inputs; 2 digital inputs)
Faders	(12) channel; (1) master L/R
EQ (per channel)	3-band mode: low shelf 40 Hz–1.5 kHz; mid peaking 200 Hz–8 kHz, Q 0.5–16 octaves; high shelf 500 Hz–18 kHz. Boost/cut (per band) ±12 dB. 2-band mode: high and low shelf
Recording Modes	Mastering (MAS), MultiTrack Pro (MTP), MultiTrack 1 (MT1), MultiTrack 2 (MT2), Live 1 (LIV1), Live 2 (LIV2)
Songs	200 per 2 GB partition
Locator Points/Markers	64/1,000
Levels of Undo/Redo	999
Analog Inputs	(2) XLR mic/line; (6) balanced ¼" TRS mic/line; (1) ¼" unbalanced, high-Z guitar (overrides input 8)
Analog Outputs	(8) RCA, independently assignable in pairs as either channel direct outs or aux A, aux B, main, or monitor outs; (1) ¼" TRS headphone
Digital I/O	Stereo S/PDIF: (1 pr.) coax; (1 pr.) optical; independent outputs only
Other Ports	MIDI In, Out/Thru; (1) ¼" footswitch; DB25 SCSI-1
Options	VS8F-2 effects expansion board (up to 2); CD-RW drive

If you want to create several different setups for tracking, a few more for mixdown, and some specialized ones for bouncing tracks, EZ Routing makes it a simple process. I created several microphone setups for the various vocalists I was working with; later, I instantaneously recalled the tracking setups to do overdubs that exactly matched the original sessions.

The VS-1680 comes with 3 read-only preset routings and 29 editable user routings. Two types of EZ Routings are provided: Templates, in which you can change the settings from a single list of all routings; and Step Edit, in which you can change the settings using a question-and-answer dialog.

You can save all current mixer settings as a Scene. Because the faders are not mechanized, the fader positions will not reflect the actual values when you recall a Scene; therefore, you can create a Scene that, when recalled, does not alter the fader values. You can even morph between Scenes over time.

If you really want to see automation at work, though, try out the unit's dynamic automation, which lets you record all mixer parameter moves and dynamically play them back in real time. An edit decision list allows you to edit the mix data (which includes effects data) so you can automate effects in real time. The mixer's controls send MIDI messages, too, so you can save your moves to an external MIDI sequencer. With all these features, the mixer section of the VS-1680 rivals many stand-alone digital mixers.

MOVING AND GROOVING

Roland provides Cut, Copy, Paste, Insert, Erase, and Move functions, which you can apply to phrases or to entire tracks. Individual phrases can be divided, trimmed (at the beginning and end), and named. You can even exchange performance data between tracks, so what was on, say, track 1 goes to track 2, and vice versa.

The VS-1680 has 64 Locator points (compared with the VS-880's 32), which you can instantly access with dedicated buttons, and up to 1,000 Markers that you step through in order. These features help you get to edit points quickly and easily. You also get loop recording,

ROLAND

VS-1680 portable digital studio
$3,195

VS8F-2 effects card $395

CD-RW SCSI drive $750

FEATURES ■■■■

EASE OF USE ■■■▮

AUDIO QUALITY ■■■■▮

VALUE ■■■■■

1 2 3 4 5

PROS: Excellent recording quality at lower compression settings. Well-designed user interface. Flexible internal routing. High-resolution I/O. Studio-quality effects on optional cards. Can create audio CD with optional CD-RW drive. Stand-alone recording system.

CONS: Audio quality suffers at higher compression ratios. No dithering on digital outputs. Cannot directly interface with MDMs. Only two XLR inputs. Manual is sometimes confusing.

CIRCLE #437 ON READER SERVICE CARD

auto punch-in/out, and manual punching via the Record button or footswitch. Thanks to the waveform display on the LCD, you can edit with precision. If you mess up, you'll be glad that Roland provides 999 levels of undo and redo.

A MANDATORY OPTION

The VS-1680 has four internal effects sends, each of which can be assigned to either the RCA aux A and aux B outputs, the direct outs, or the two optional internal effects cards. Each VS8F-2 card contains two stereo, 24-bit effects processors. You can add the optional effects to specific channels by inserting them between the channel EQ and fader, or you can add them to the overall mix before the master fader.

The VS8F-2 card offers a multitude of effects that were gleaned from the best of the Roland and Boss effects processors. I could do a whole review just on this card, but space limitations allow only a few sweeping comments.

You get a variety of the expected reverb, chorus, delay, dynamics processing, flanging, phasing, and multi-effects algorithms, all of which are detailed and user-editable. Surprisingly, the VS8F-2 also contains mic simulator, lo-fi processor, Voice Transformer (to provide independent pitch and formant controls), hum canceler, vocal canceler, tape echo, vocoder, and RSS (Roland Surround Sound) effects.

The VS8F-2 card has many of the modeled guitar preamps, cabinet simulators, and guitar effects found in the Roland VG-8 and Boss GT-5 guitar processors. All you need is your guitar and cable, connected to the VS-1680's high-Z guitar input, to create admirable guitar tones with remarkable ease. Evaluating guitar tones is extremely subjective, so I'll just say that the VS-1680's guitar section functions as well as Roland's other pedals, multi-effects boxes, and simulators—if you like those, you'll like the implementation in the VS-1680. Patch in a favorite pedal or preamp for a little added flavor, and you have some serious guitar recording at your fingertips.

The effects card's compressors are much like those of most other digital dynamics processors. I'm not fond of digital compressors; I generally have difficulty getting good results with them. The VS-1680's compressors are adequate, however, especially if you have time to tinker with them and work out exactly the right settings for each instrument. This is usually hard to do in a live vocal session, so I used Empirical Labs' EL8 Distressors on the vocals and bass. I used the internal compressors for tracking keyboards, though, and was pleased with the results. Overall, I wouldn't depend on them for high-performance compression, but they are functional.

Except for the compressor/limiter, the effects are outstanding. I can't say enough good things about them. If you get a VS-1680, go ahead and buy a pair of these effects cards; you won't regret it.

BUS TRANSFERS

Before I left for the road, my first task was to transfer two partially completed songs, which were originally recorded on ADATs, into the VS-1680 for later overdubs and mixing. I wanted the transferred tracks to maintain sync with my original MIDI sequence and ADAT locations, in case I needed to redo any synth or sampled parts.

I hit the road, hoping that the VS-1680 wouldn't let me down.

A Panasonic MDA-1 ADAT generated the time code, and I was careful to match the time-code settings in the VS-1680 (which supports all time-code types and rates). There is no word-clock input on the VS-1680, so it must slave to either a digital input signal or MIDI Time Code (MTC). The unit runs on a 24-hour time line, so if no offset is introduced, the transferred tracks follow the same time numbers as the master. That made my task easier.

Unfortunately, there is no way to simultaneously transfer eight tracks digitally between the VS-1680 and an MDM, so the only option is to transfer two MDM tracks at a time via the S/PDIF coax or optical digital inputs. (The VS-1680 *can* simultaneously output different track pairs through the two digital outputs, allowing four tracks to be transferred to another device, as long as you can deal with using both coax and optical S/PDIF.)

Of course, it's possible to use the eight RCA connectors (which have 20-bit A/D converters) for analog transfers; due to time constraints, that is the method I chose. Even with the extra A/D and D/A conversions, the prerecorded tracks sounded great. True, I had to sync the VS-1680 via MTC, which is not sample accurate, but it was good enough for my purposes.

USING V-TRACKS

While on the road, I made extensive use of V-tracks, not only for recording multiple performances of vocals, drums, bass, and guitars but also to create several arrangements of the same song in different keys. Because the record contained several guest artists (Dionne Warwick, Grant Geissman, and John Pagano, among others), I wanted to be sure that I had alternative arrangements available in neighboring keys.

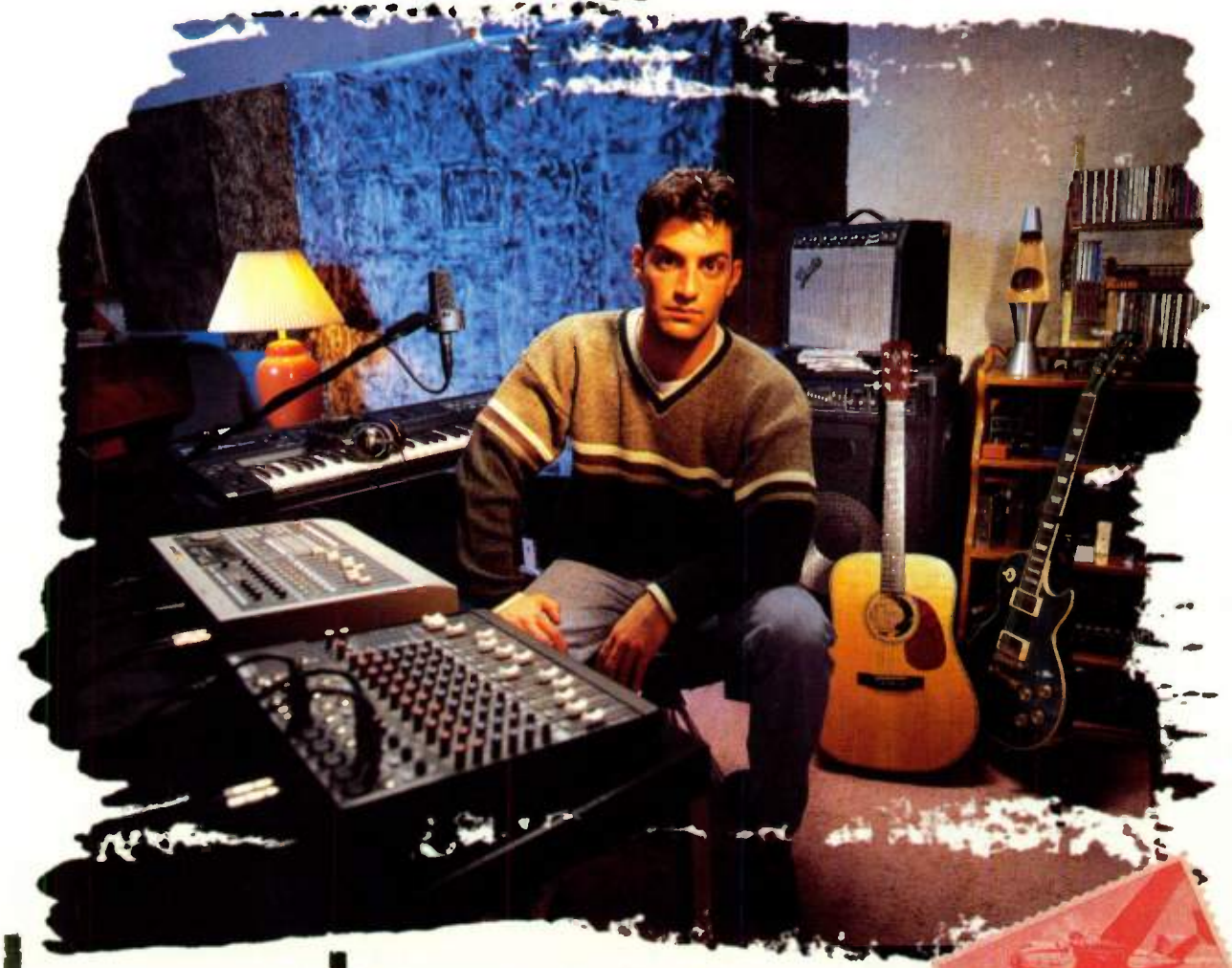
Most of the singers were accompanied by MIDI tracks, with only minimal overdubs at this stage of the project. Of course, transposing would have been easy with MIDI tracks, but I couldn't always have my MIDI rig available, so I had to record those parts on the VS-1680 as I went.

Fortunately, the virtual tracks gave me some of the flexibility I would have had with MIDI tracks. I used V-track 1 on all channels for recording in the originally intended key. Then I created two alternative versions: the bass, keyboards, and

Recording Time (2 GB partition)

Mode	48 kHz	44.1 kHz	32 kHz
MTP	742 min.	808 min.	1,114 min.
MAS	370 min.	404 min.	556 min.
MT1	742 min.	808 min.	1,114 min.
MT2	990 min.	1,078 min.	1,484 min.
LIV1	1,188 min.	1,292 min.	1,782 min.
LIV2	1,484 min.	1,616 min.	2,228 min.

If I wanted a hobby,



I would have picked stamp collecting.

Recording isn't my hobby, it's what I do. And even though I do it at home, I still expect the sound to be dead on. So when the vocal tracks weren't cutting it anymore, I upgraded my mic to the AT3525. You wouldn't believe what that studio condenser does for my sound. Now the old ball mic is just collecting dust.

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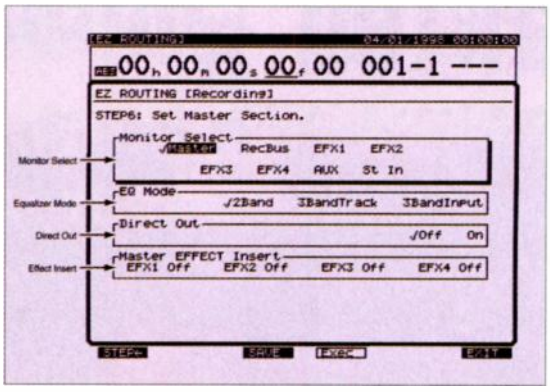

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FIG. 2: The EZ Routing feature greatly simplifies the process of creating recallable setups. Shown here is only one part of a routing that is optimized for recording.

other tonal instruments were transposed a semitone up on V-track 2 and a semitone down on V-track 3. Two singers needed to change keys, but that only meant switching to different V-tracks. Within seconds we were ready to record in the new key. Amazing.

V-tracks are great for slamming down multiple takes of vocals. Considering that a lot of these tracks were being recorded in hotel rooms at one o'clock in the morning, after gigs or cross-country flights, I just let the singers do their thing and collected several takes on V-tracks. Later, I worked with these tracks to create composite performances.

GETTING IT DONE

One afternoon, between matinee and evening performances, I used the VS-1680 to record drummer David Crigger in the beautiful Seattle Symphony Hall. As with everything else on this project, we had to work quickly. The VS-1680 has only two XLR inputs, so the biggest challenge was to get a collection of mic cables patched into the unit's mixer. After collecting an assortment of adapters, I set up the VS-1680 next to Crigger and fed each of us a headphone mix. We were able to set up, record drum tracks for three songs (with a great room sound), restore everything to the setup for the show, and have dinner—all inside of three hours.

After a series of overdubs, it was time to mix, and I was pressed right up against my deadline. By this time I had confidence in the fidelity of the VS-1680, so I elected to mix entirely in the workstation. Thanks to the quality of the effects on the VS8F-2 cards, I had no problem creating the desired ambience, delay, and EQ effects for the mixes. For some mixes, I used automation; others didn't require it. I dumped the mixes to a computer for editing and sequencing before creating a master CD-R, but I could have done it all with the VS-1680 and the optional CD-RW package.

FINAL THOUGHTS

Considering how portable, yet powerful, the VS-1680 is, it is difficult to keep a perspective on how much value is packed into this little tabletop workstation. I can't possibly cover all of its features in detail, so you'll have to check out the unit yourself to get the lowdown on phrase editing, effects inserts, tempo maps, the internal metronome, audio scrubbing, Vari-Pitch, and synchronizing additional VS-series recorders.

The VS-1680 has some weaknesses, too. I wish there were dithering on the 24-bit digital outputs when connected to a 16-bit recorder, and I would like a way to digitally transfer eight tracks at a time to an MDM. I would also prefer more than two XLR inputs, and I *really* wish that Roland would pay an undergraduate of English to proofread its documentation. (According to the company, a new manual should be available by the time you read this.)

The bottom line is that I was able to accomplish almost everything I needed to do with the VS-1680. The recorder sounds excellent, it is easy to operate, and it provides lots of control over the entire recording process. The line between high-end and personal studio recording gear has become so blurred that it really comes down to whether you are recording something worth recording, and whether the person in the driver's seat is experienced and has good ears.

In any case, the VS-1680 probably won't hold you back in the recording process. Roland has combined sonic excellence with flexible ease of use in a winning package for pros and hobbyists alike.

Composer and producer **Rob Shrock** *is the musical director for Burt Bacharach. He has also worked with Elvis Costello, LeAnn Rimes, Luther Vandross, Chrissie Hynde, Stevie Wonder, and a host of other artists.*

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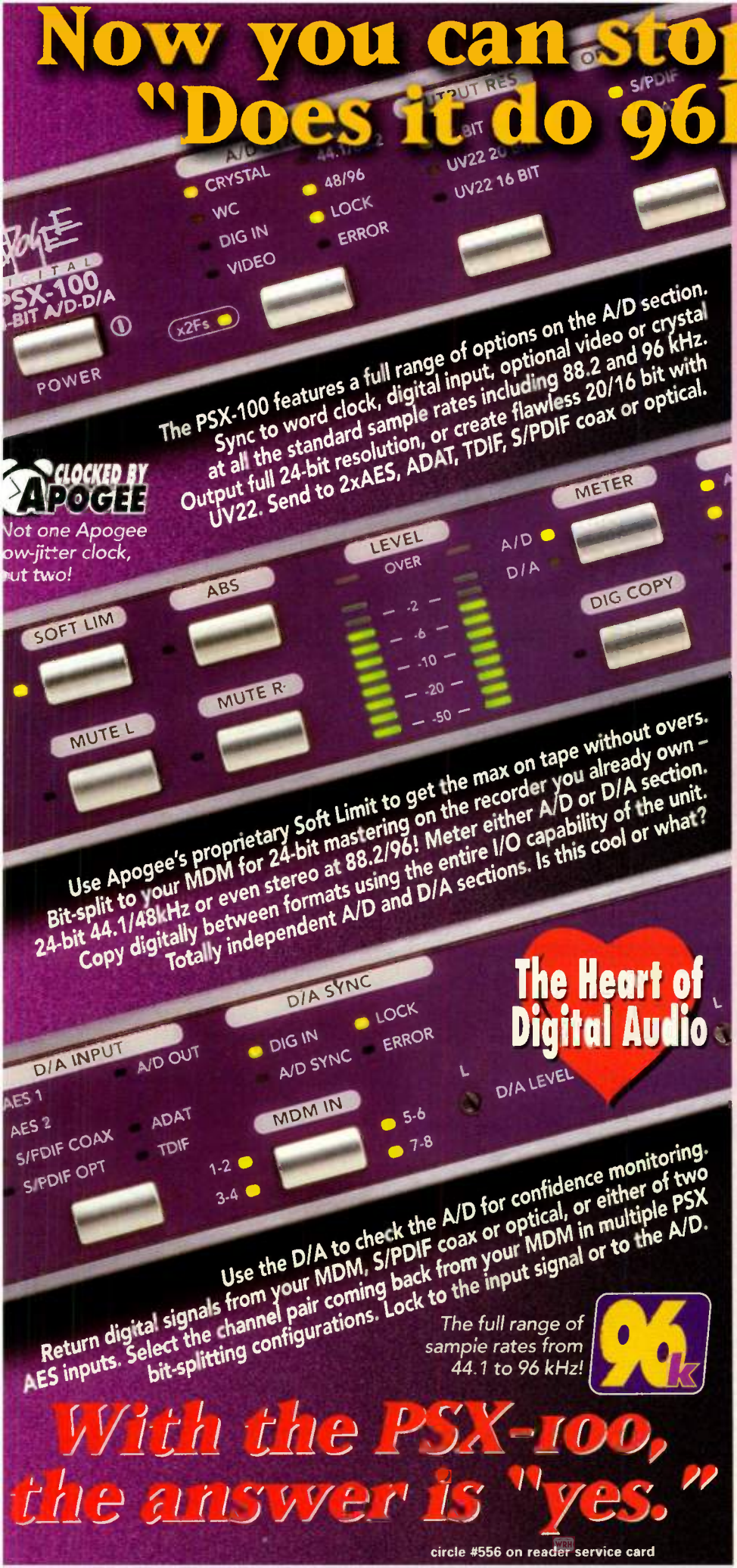

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SEK'D

SAMPLITUDE 2496 5.12 (WIN)

Digital audio software with more than a few extra bits.

By Zack Price

Although the 16-bit digital audio format isn't dead yet, it is rapidly being replaced in professional audio applications by 20- and 24-bit formats. Moreover, standard sampling rates—especially for mastering purposes—are increasing from 44.1 kHz to 96 kHz.

Samplitude 2496, the new "senior member" of SEK'D's *Samplitude* family of audio editors, supports a number of higher resolutions and sampling rates—up to 24-bit at 96 kHz (hence the "2496" designation). The company, however, has done more than merely increase the resolution in its top-of-the-line product: it has also added important new features and revamped key aspects of the program. Furthermore, SEK'D markets several hardware products that take advantage of *Samplitude 2496*'s new and improved capabilities (see the sidebars "Conversion Experience" and "Card Sharp").

SOMETHING OLD—AND NEW

One thing that hasn't changed in *Samplitude 2496* is the basic structural hierarchy also found in earlier versions of the program. As before, the central workspace is the VIP, or Virtual Project, window (see reviews of *Samplitude* in the March 1996 and January 1998 issues of **EM**). The VIP is a multitrack window that contains graphic representations of digital audio data known as Objects (see **Fig. 1**). It is where you assemble data, perform mixes, and undertake most of your editing and processing tasks.

You can create Objects, which are simply pointers to actual audio on your drives, in several ways. They can be created from recordings you've made directly in the VIP window onto a single track or, provided you have multiple stereo cards or a multichannel sound card, onto multiple tracks. You can also create Objects using data from Hard Disk Projects (HDPs) or RAM Projects (RAPs). The difference between these two is simply that one records directly to your hard drive, and the other records into RAM. (Note that whenever you record an Object directly into the VIP window, a corresponding HDP is also automatically created.) After recording, you simply highlight all or parts of the data in the HDP or RAP window, drag it onto the desired track, and position it in the VIP window.

Keep in mind that RAP data is played back directly from RAM, resulting in less wear and tear on your hard drive and giving you faster access to the data. However, you'll be limited to the amount of free RAM you have available. Therefore, RAPs are most useful for recording short sections of sounds that might be used repeatedly, such as rhythm loops, short musical phrases, or sound effects.

Regardless of how you record, you'll find that *Samplitude 2496* has refined the procedure for creating recordings when compared with earlier *Samplitude* versions. For instance, you no longer have to create separate VIPs for monophonic and stereo tracks; a single VIP can now contain both types of tracks. You can also pan mono tracks, which was not possible previously. Furthermore, you can directly assign a track to be recorded or played back through the left or right channel of a sound card. This is especially useful for those using slower computers and multichannel cards with I/O routed through an outboard mixer. Rather than burden the computer by using *Samplitude*'s virtual mixer, you can simply employ the pan, EQ channel inserts, and other features of an external mixer. This conserves system resources, and that can translate into more tracks or additional real-time effects (more details later).

Samplitude 2496 supports several resolution options that you set either in the Global Record Options or from within the File menu. The choices for data are 16-bit or Floating Point. The 16-bit option is self-explanatory; Floating Point is especially well-suited for users with 20- and 24-bit cards. Floating Point can also be useful for musicians using 16-bit cards, however, because the program will process real-time, offline, and DirectX effects with up to 32-bit internal precision when this feature is active.

Sampling rates, on the other hand, are set ahead of time in the VIP Setup window, or from within the recording setup window for Hard Disk and RAM Projects. No matter where you set the resolutions and sampling rates, be sure that your settings are consistent. *Samplitude 2496*, like most digital audio programs, can't play files with differing rates and resolutions simultaneously.

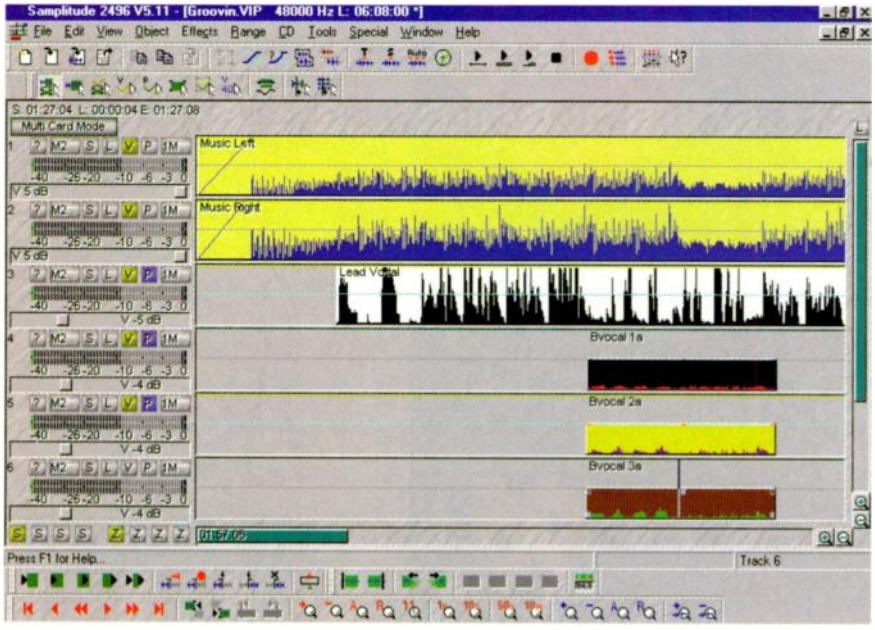

FIG. 1: The main work area in *Samplitude 2496* is the VIP window. Digital audio appears here as Objects and can be manipulated in a variety of ways. Volume and pan can be drawn directly on the waveform displays.

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MOUSE HUNT

Another feature that distinguishes *Samplitude 2496* from the older versions is its enhanced and more accessible Mouse Modes, which define how the mouse will function when you are in an editing session. Previous versions of *Samplitude* provided a variety of mouse behaviors, but you had to right-click to view or change the current Mouse Mode option. With the addition of Mouse Mode buttons, users can quickly see which mode they're working in and switch easily from one mode to another. The right mouse button is now free for opening context-sensitive editing menus.

The default Mouse Mode in *Samplitude 2496* is the Universal mode. When Universal is enabled, you can highlight a range of audio data by dragging in the upper half of the Object's display, or move and select the Object by clicking in its lower half. However, you can override this behavior by enabling either the Volume or Pan buttons that appear to the left of each track (see **Fig. 1**). This makes it easy to perform simple volume and pan adjustments without switching modes.

Other Mouse Modes are dedicated to specific tasks, including Range Only mode, which limits the left mouse button to highlighting ranges. This prevents Objects that haven't been locked from being accidentally moved from their original positions. An Object mode is provided for selecting, moving, and resizing Objects. Resizing an Object changes the amount of audio it will play back.

Curve mode limits the mouse to creating or adjusting linear volume and pan curves—you simply drag on the volume or pan "handles" that appear directly on the Object. (You can also change the overall Object volume by moving the middle handle up or down.) Volume Draw and Pan Draw modes allow you to create freehand drawings for the type of curve you want. However, the Volume or Pan buttons at the left of the corresponding track must also be active before any curves can be drawn within the Object.

Although I find it easier to use the new Mouse Mode buttons in *Samplitude 2496*, users of previous versions may prefer to work with the mouse the way they're accustomed to. For them, a Version 4.0 button makes the mouse function just as it would in the earlier version. I would encourage people to learn how to work with the various mouse buttons, though; it's not difficult, and the advantages gained from using the context-sensitive menu are worth the effort.

Samplitude 2496
Minimum System Requirements
Pentium 60; 16 MB RAM (32 MB if running under Windows NT); Windows 95/98/NT; full-duplex sound card; CD-ROM drive for installation

SOME LIKE IT HOT

Samplitude does such a good job of letting users navigate with a mouse that you might forget that most of the program's functions can also be carried out using hot keys (single or combined keyboard strokes). *Samplitude*

2496 includes an extensive default list of hot keys, but strangely enough, this list is not as large as in previous versions of the program.

SEK'D has also changed some hot-key assignments, which may come as an unpleasant surprise to previous users. You can, however, use the Hot Key Assignment Editor to add to the list of hot-key assignments or to alter the default assignments into whatever keystrokes make sense to you. But keep in mind that if you alter the defaults, they will no longer correspond to the hot-key prompts listed in the program's menus.

Similarly, you can now customize the toolbars to display only those buttons that you wish to use. You can also change the buttons' appearance from a raised look to the slightly flatter "touch membrane" style shown in **Figure 1.**

RECORD AND PLAY

In addition to the new features described above, *Samplitude 2496* includes other enhancements to its recording and playback arsenals. One useful feature is the automatic sample-rate conversion that you can apply during recording. Send the program a 48 kHz signal from a DAT, and you can end up with a 44.1 kHz file on your drive that is suitable for CD burning.

Speaking of CD burning, *Samplitude 2496* includes all the tools you need for creating professional CD masters. You can even apply automation on multiple tracks while a disc is burning. That saves you from having to create an image file on disk before making your CD.

Another new feature is punch-in and punch-out recording on multiple tracks. You can easily punch in and out on the fly, or if you prefer your punch points to be more precise, just use the appropriate buttons to set in and out markers. A button is provided to clear the punch markers.

Among the program's playback features is the handy ability to adjust the playback rate of a file in real time. Unlike the way in which this function is often implemented on varispeed CD players, the output sample rate doesn't change. That means that if you're sending your output digitally to a device like a DAT, that device will stay synched to your signal. *Samplitude 2496* also is able to trigger a MIDI or AVI file when you start playback of a project. (Additional MIDI editing options should be available in the program by the time you read this.)

Samplitude 2496 has a very interesting new performance feature: the Live Input mode. When this mode is activated, you

SEK'D
Samplitude 2496 5.12
audio editing software
$799

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

PROS: Exceptional number of features and functions in one package. Extensive DirectX usage and support. Includes external MIDI device control.
CONS: Doesn't support SMDI.
CIRCLE #438 ON READER SERVICE CARD

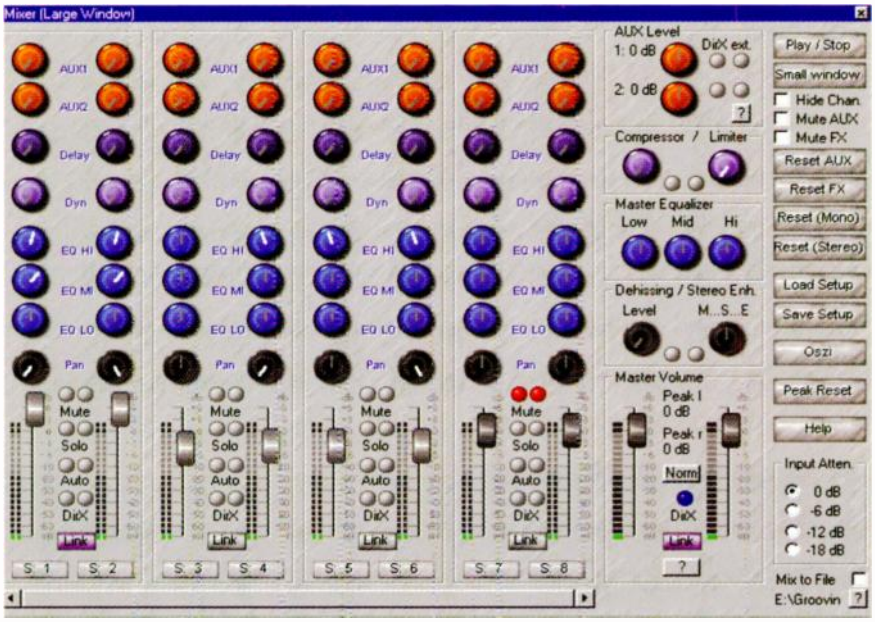

FIG. 2: The Mixer window now lets you simultaneously access DirectX plug-ins from within each track, auxiliary sends, and the Master Volume section.

can use all of the program's mixing functions, including DirectX effects, to mix live inputs from sound-card input channels. Live Input mode can be used, for example, to add effects to tracks you're transferring from an MDM into *Samplitude 2496*. You could also create a stereo mixdown of the live input by selecting the Mix To File option found in the lower-right corner of the Mixer window.

Additionally, you can use the Live Input mode to turn *Samplitude* into a live mixer and effects processor in a pass-through mode (without recording to hard disk). This lets you take advantage of DirectX and *Samplitude*-native real-time effects (like its FFT Filter and Multiband Dynamics processors), in conjunction with outboard tracks and analog boards, in real time—well, almost real time, anyway. Audio processing in a host-based digital audio program produces delays at the output stage compared with the input stage. Depending on *Samplitude*'s buffer settings, this latency can be from 200 to 700 milliseconds.

Latency is not an issue if you are simply transferring multiple tracks at once from an outboard deck or MDM, because all tracks are delayed by the same amount. However, you may experience problems synchronizing live input from other sources with tape sources not routed through *Samplitude 2496*. In some cases you can get around the problem, provided you have MDMs that can shift tracks far enough backward to compensate for the delay caused by the Live Input mode. Naturally, you need to make sure to shift backward only those tracks that aren't routed through *Samplitude*.

DIRECTX-FILES

Another exciting feature of *Samplitude 2496* is its support for DirectX plug-ins. The new version allows you to use these plug-ins in very flexible ways. For instance, when the Mixer window is open in a VIP (see **Fig. 2**), you can use DirectX plug-ins in real time at the Master Volume section, on individual channel inserts, and at both Auxiliary returns. Furthermore, you can activate these access points simultaneously, if you've got the computer power to do so. It's also possible to chain plug-in effects together, subject only to the limitations of your computer's performance. Just be sure to chain the effects in a logical order. Also, watch your gain settings from one effect to the next, as you would for outboard effects processors.

Samplitude 2496 gives you the ability to perform real-time previews of DirectX

CONVERSION EXPERIENCE

Having software with 24-bit, 96 kHz sampling ability is useless unless your hardware can convert analog signals to 24-bit digital audio. Fortunately, SEK'D markets a little box that is your entry to high-end, high-resolution recording and playback. The ADDA 2496 S ($999) is a half-rack stereo A/D/A converter; as its name implies, the unit is designed to convert analog signals to digital audio at 96 kHz with up to 24-bit resolution. In fact, the A/D converter samples at 44.1, 48, and 96 kHz with up to 24-bit resolution, while the D/A converter outputs audio at 44.1, 48, 88.2, and 96 kHz.

The audio and digital connections on the back panel are very straightforward. The analog I/O uses balanced jacks, while the digital I/O offers a choice between AES/EBU connectors (XLR type) and S/PDIF coaxial (RCA type) connections. The unit also comes with its own internal power supply, so there's no bulky wall wart to contend with.

The ADDA 2496 S is simple to operate. The device uses two toggle switches in four different combinations to set the input sampling rates. It also includes a toggle switch for setting the copy-protect flag (as if anyone ever uses that feature), and another toggle switch for selecting between the AES/EBU and S/PDIF digital audio jacks. In addition, the front panel has three green LEDs to indicate the current sample rate, and a red LED lights up whenever there is a sampling mismatch or locking error between the ADDA 2496 S and another digital audio device.

When recording, SEK'D recommends that you set the input control knobs all the way down to -10 dB to avoid clipping. When I ran the analog outputs directly from my DAT deck into the ADDA 2496 S, I noticed that the clip indicators would light occasionally, even at this lowest setting. Likewise, the recording meter's clip indicator in *Samplitude 2496* would light up. My advice is to watch your levels and be certain that you can control the sound source before routing it into the ADDA 2496 S.

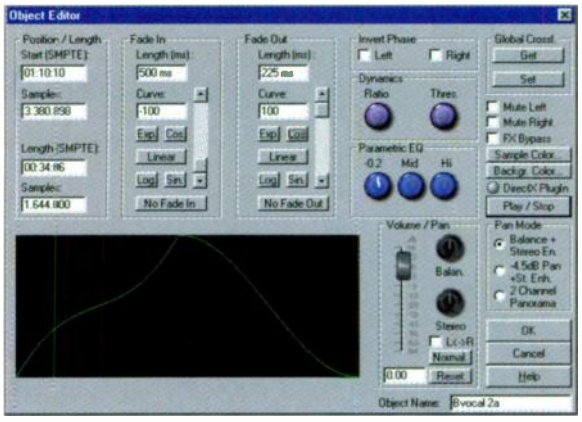

FIG. 3: Objects can be individually processed using the Object Editor. This is particularly useful for adding effects to a specific section of a track in the VIP window.

effects in HDPs and RAPs and then destructively apply those effects. However, the most interesting use of DirectX plug-ins can be found in the program's Object Editor (see **Fig. 3**), which itself deserves special recognition. As mentioned earlier, Objects are graphic representations in the VIP window that point to digital audio data contained in HDPs or RAPs. With the Object Editor, you can nondestructively apply effects and processing to specific Objects in a track without affecting other Objects in that same track.

Let's say, for example, you want the lead vocal to shift into "telephone voice" for one verse and then shift back to normal. Simply create an Object consisting of the section of the vocal track you want to change. Select that Object and open up the Object Editor. You can now apply *Samplitude*'s own parametric EQ settings, or use a DirectX EQ plug-in, for just that Object.

I have just one caveat about working with the Object Editor. In the previous example, it is likely that the lead vocal track would be recorded in the VIP as one long Object. To apply the special effect in the Object Editor, you'd need to split that single Object into three parts: before, during, and after the effect. *Samplitude* lets you easily split Objects, but it retains the original Object name for all the new Objects. You must give a unique name to any Objects to which you apply effects, because the effect will be applied to all Objects with the same name.

A final word about DirectX plug-in operations in general: When in Mixer mode, *Samplitude 2496* works with most DirectX plug-ins if they can process the audio immediately and completely (with no buffering) and without changing the length of the processed material. When in Effect mode, there are no processing restrictions on buffering and time alteration. Moreover, the quality of the effects depends on *Samplitude*'s internal precision settings. Some plug-ins may work only at 16-bit resolution, which makes them unusable when operating *Samplitude 2496* in 32-bit floating point resolution.

You'll also need to have DirectX 6.0 installed on your system. (This is not an issue for Windows 98 users.) Fortunately, the DirectX files you need are

included on the *Samplitude* CD-ROM and can be installed from there if needed. Be sure to remove the previous version of DirectX from your system before installing the new package. Simply updating previous versions of DirectX with the DirectX 6.0 update file (roughly 1.5 MB) is insufficient for use with *Samplitude 2496*. (For more information, go to www.microsoft.com/windows/downloads /default.asp and download ENDUSER.EXE.)

NOT FADE AWAY

You can now control *Samplitude* with an external MIDI control surface such as the Peavey PC 1600x MIDI fader unit and the MIDI Control Panel application (see **Fig. 4**). External device control is a major advance for host-based hard-disk recording programs, because user interaction is no longer limited to mouse or keyboard input for editing and control functions. Users can now perform simultaneous tasks such as manual channel crossfades, which would be impossible with mouse input alone. Best of all, you don't have to invest in an expensive, dedicated DSP-based system to get this functionality.

By default, *Samplitude 2496* is coded to respond to the MIDI parameters of Preset #11 (Pro Tools Vol/Pan) in the Peavey 1600x. (You can, of course, have it "learn" the data structures of your own hardware.) Depending on the window or mode, the faders and buttons on your external device control different features in *Samplitude 2496*. For instance, when in Mixer mode, Peavey faders 1 through 8 control volume for tracks 1 through 8; faders 9 and 10 control master faders; 11 through 13 control the three master EQ bands; 14 and 15 control aux 1 and aux 2 sends, respectively; and fader 16 controls pitch adjustment. The hardware buttons control track mute and solo functions, track and master fader links, DirectX activation, scrubbing, and playback stop/start.

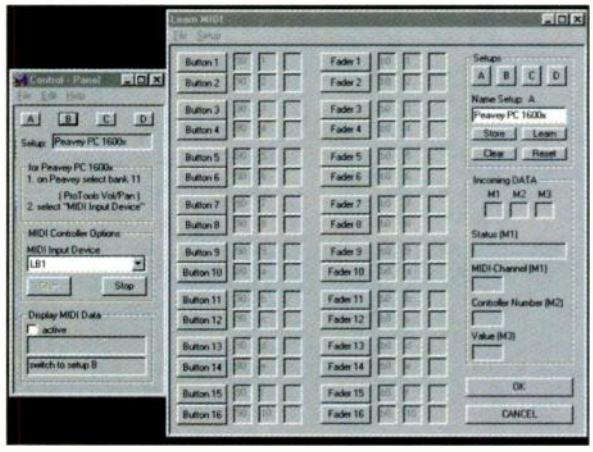

FIG. 4: You can now control various functions in *Samplitude 2496* via MIDI, using the Control Panel program shown here and a control device such as the Peavey PC 1600x MIDI fader unit.

In *Samplitude*'s VIP mode, the 1600x faders control such features as zoom in/out, scrolling, cursor positioning, and range setting. The buttons perform functions such as incremental zoom in/out, cursor positioning on ranges, playback start/stop, cursor position at playback stop, and scrubbing when used with fader 16.

I found working with an extended control interface a joy, especially after being limited to mouse and keyboard input for so long. SEK'D keeps updating this useful utility, and you can head to the company's Web site to get the latest version.

OVER AND OUT

Samplitude 2496 has many new and improved features beyond those I've mentioned. For instance, the program now supports the Yamaha DSP Factory system, as well as some EIDE CD-Rs for CD burning. Similarly, *Samplitude 2496* can extract digital audio from certain EIDE-based CD-ROM drives; previous versions supported only SCSI-based CD-Rs and CD-ROM drives for these functions. Again, check the SEK'D Web site for an updated list of supported devices.

Moreover, SEK'D has been working to improve the quality of its documentation, and it shows. There's an entirely new manual that is very clear and geared toward American users of the program. I applaud SEK'D's efforts to be responsive in this area.

Despite the program's extensive list of features, some enhancements are still lacking, such as support for SMDI (SCSI) sample dumps instead of the slower SDS standard. This would make *Samplitude 2496* a complete solution for all digital audio editing needs.

Overall, though, *Samplitude 2496* is an outstanding program. In fact, I think it's the best host-based hard-disk recording software for Windows. I strongly encourage you to check it out.

Zack Price *would like to thank Steve Henson for keeping him in Split Enz.*

CARD SHARP

SEK'D also markets the Prodif 96 ($799), a PCI-based stereo digital audio card that is a perfect mate to the ADDA 2496 S. The Prodif 96 supports sample rates of 32, 44.1, 48, 64, 88.2, and 96 kHz at up to 24-bit resolution. The card can also output analog audio at the aforementioned rates, but at 20-bit resolution—which is still more than adequate for most monitoring situations.

The Prodif 96 connects to external devices via optical S/PDIF (Toslink), coaxial S/PDIF (RCA type), and AES/EBU (XLR type) digital audio connections. The latter two options are available on breakout cables attached to a 9-pin D-connector on the card's backplate. Unfortunately, these cables aren't marked. I managed to find out that the red RCA jack is S/PDIF out and the white jack is S/PDIF in. Likewise, the male AES/EBU jack is digital out and the female is digital in. An addendum to the manual is now included with each sound card to identify the ins and outs of the breakout cables, but hopefully SEK'D will also label the next batch of cables.

The Prodif 96's analog output uses a ¼-inch TRS jack and can be connected using an insert send/return cable. You can plug headphones into this jack, using the switch next to the jack to change the impedance level to drive the headphones. The card also has an internal digital audio input that connects to the digital audio output of a CD-ROM drive.

The Prodif 96's latest drivers include ASIO drivers for use with Steinberg's *Cubase VST* digital audio sequencing program.

Sound Design.

Incredible Sound

The Yamaha CS2X is filled with phat, fresh sounds for today's contemporary music scene. It boasts 64-note poly, a huge 16MB wave ROM, and true 24 dB resonant filters per voice. Its multi-effects go way beyond the typical, with 12 reverbs, 14 chorus/delays and 62 variation effects like Rotary Speaker, Distortion and Touch Wah. In performance mode, you get 4-way splits and layers with customized effects sets, plus 12 MIDI channels of additional sound for backup. In multimode, CS2X is a full 16-part multitimbral powerhouse.

Dynamic Design

The CS2X's slick silver finish and translucent blue knobs stand out in a crowd of look-alike synths. But design is more than just a pretty face: Yamaha created the CS2X to make it especially easy for you to create great music. The eight analog-style knobs – for direct editing of the most important synth parameters – give you intuitive, real-time control; your next killer sound is just a knob twist away. Additional design features include scenes that let you easily store and recall variations of a voice in realtime. And a MIDI-syncable arpeggiator adds style to your music.

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ALESIS

STUDIO 32

This flexible console lets you mix down without getting mixed up.

By J.J. Jenkins

I see a lot of equipment come through my studio, some of which is very complex. At times it gets tiresome trying to decipher and remember all the various access codes and pathways to additional layers and functions. Not that I'm against technology; a quick walk through my studio would quickly convince anyone otherwise. But I find it refreshing to run across a piece of equipment that is fairly straightforward and easy to use. That's why I was delighted when I pulled Alesis's friendly-looking, compact Studio 32 out of the box.

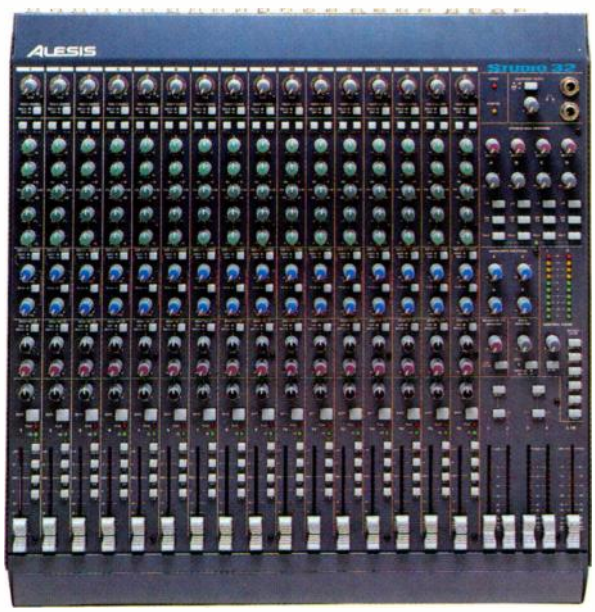
The Studio 32 may be straightforward to use, but it certainly isn't stripped down. This console is full of features. It boasts enough faders and knobs to impress the seasoned engineer—and to be slightly daunting to the novice. Have no fear, though: mixers usually look more complicated than they are, and the Studio 32 is no exception.

GETTING INTO CHANNELS

Each of the board's 16 input channels has an XLR mic input; a balanced, ¼-inch, line-level input; and a balanced, ¼-inch "tape return." (Of course, the tape return could just as easily be called a "DAW return" these days.) The mic and line inputs are not simultaneously available, so you should connect one or the other, but not both, for each channel (see **Fig. 1**).

The beauty of the Studio 32's tape returns is that they are wired internally and can be routed with the flick of a switch either to the main input channels (where you want to send them for mixdown) or through the monitor 1/2 section for tape playback and monitoring while recording. Not having to repatch to hear the playback is always a plus to me.

The tape inputs are made even more versatile by a button on the monitor 1/2 section that can switch between the tape return and the mic and line inputs. That means you can monitor the recorded sound coming from the tape deck or the live inputs—again without repatching. So the Studio 32 is actually a conventional 16-channel board with 16 *additional* inputs that can be used in a multitude of configurations—hence, the product's name.



Flexible routing, plenty of inputs, clean sound, and ease of use are the hallmarks of the Studio 32 recording console from Alesis. Although it's a 16-channel mixer, you can have up to 40 inputs at mixdown.

The mixer's ¼-inch channel-direct outputs let you route your sound source straight to the recording deck. These direct outs are postfader, so any fader adjustments that you make will be represented on tape.

Except for the mic inputs, most connections are ¼-inch TRS (see the table "Studio 32 Features") and will work with balanced or unbalanced connectors.

CRUISE THE STRIP

Let's go through the components of a Studio 32 channel strip. At the top is a trim pot that adjusts the mic or line signal level coming into the channel. A fader-source button below the trim knob switches the input from mic/line to tape monitor. The trim can be used only with the mic/line input and has no effect on the tape return. (The tape inputs work with +4 dBu or -10 dBV signals and don't require trim pots.)

The equalizer is very nice; I think it's one of the best features of this board. Unlike the EQ in some mixers—especially British consoles—the Studio 32's EQ sounds virtually transparent and doesn't add a distinctive personality.

The high- and low-band controls are pretty much equivalent to bass and treble adjustments, but the midrange is a real parametric equalizer. If you've worked with a parametric EQ before, you know how useful the ability to fine-tune the frequency, level, and bandwidth can be. Having a sweepable midrange is nice, but a fully parametric midrange band on each channel is what you need for professional recording. A switch lets you bypass the EQ.

A button enables an 18 dB/octave, low-cut (highpass) filter that rolls off frequencies below 75 Hz. It's great for getting rid of low, rumbling noises when you're trying to track vocals or guitar—but don't use it on a bass or kick-drum track or you'll be in for a rude surprise. The low-cut filter works independently of the EQ.

The sends are up next. Monitor 1/2 is a stereo, prefader, premute monitor send. As I noted earlier, a source switch routes the tape input or mic/line inputs to the monitor buses. A single level knob determines how much signal goes to the master monitor 1/2 bus, and a black Pan knob controls the balance between 1 and 2.

Two blue knobs set the levels for all four mono, postfader auxiliary sends. One knob controls auxes 3 and 5, the other auxes 4 and 6. The To 5/6 aux-assign switch determines whether the level knobs control aux sends 3 and 4 or 5 and 6. A Source button lets you choose whether auxes 3 and 5 carry the signal from the channel fader or the signal from the monitor 1/2 bus (see **Fig. 2**). Being able to send the monitor signal to an effects processor via sends 3 and/or 5 is especially useful if you plan to use monitor 1/2 as a separate submixer or want to monitor with reverb. The four auxiliary sends don't have to be effects sends: they can also be used for setting up extra headphone mixes, for example. Just keep in mind that they are postfader.

For manufacturer information, please see Contact Sheet, p. 163.

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At the bottom of the Studio 32's channel strip we find a pan knob, a mute button, two LEDs, a solo button, assignment switches, and the fader. Each of the two LEDs serves a dual purpose: the red peak LED also lights up when the mute button is on, and the green -20 dB LED (which indicates that a signal of -20 dB or higher is present) also glows when the Solo button is engaged.

You can choose where to route the signal with the mixer's three assignment switches. The L/R switch directs the track to the master fader. The other two buttons send the signal to the four subgroup faders, which are assigned as odd/even pairs (1/2 and 3/4).

Far too often, I've used faders that seemed like toys. If you're doing a lot of mixing, it's important that the faders feel substantial—and the Studio 32's do.

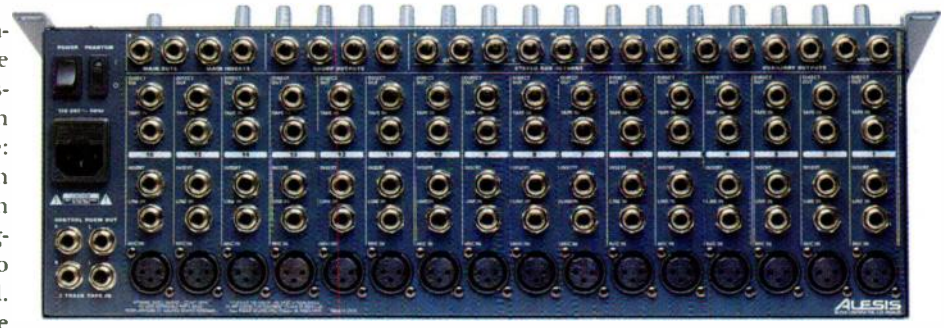

FIG. 1: All the Studio 32's inputs and outputs (except the inserts and aux returns) are balanced but also work with unbalanced lines. You get mic, line, and tape inputs on every channel; however, the mic and line inputs are not simultaneously available. Thanks to extensive internal switching, the board enables you to route signals in more ways than you'd guess by looking at the jack field.

THE MASTER REPORT

The Studio 32's master section consists of the stereo aux returns, the aux-send master controls, the group master controls, control-room features, and assorted odds and ends (such as headphone controls). Let's start with the auxes.

I really like this board's auxiliary returns. Four stereo inputs bring the total input count to 40—which is a lot for a mixer of this size. If you're returning a mono source, plugging it into the left return jack bridges the signal to both sides of the stereo return; plugging it into the right return just sends it to the right return jack.

The L/R Assign switch routes the returns to the main left/right bus, and the two Group Assign switches send the returns to the stereo subgroups (returns A and B go to groups 1 and 2, and returns C and D go to groups 3 and 4). Four master level controls determine the amounts of signal sent to the assign switches. Four additional level controls send the signal straight into monitor 1/2. The returns can also be soloed.

The aux-send master controls include four master-volume knobs, a master level control for monitor 1/2, and a Link To L/R switch that routes the monitor 1/2 signal to the master L/R fader without affecting the monitor

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feed to the headphones, aux 1/2 outputs, and control-room source switches. That's how you turn the Studio 32 from a 16-channel board (with an inline monitor section) into a 32-input board.

GROUP CONTROL

The Studio 32's master left/right bus is controlled with a single fader that handles both sides of the stereo signal. There are also four group faders. Groups 1 through 4 can be used as stereo sets of 1/2 and 3/4, with a Mono switch for each pair that center-pans the pair. So, for example, the Mono switch for group 1/2 center-pans groups 1 and 2; when Mono is off, group 1 is panned hard left and group 2 hard right. Either way, you still have individual control over each group.

A pair of switches labeled To L/R allows you to bring each pair of groups into the main mix so that you can, for instance, control multiple channels from one pair of faders. Let's say you have achieved a great vocal balance on four channels and want to control this section's overall level. You would first turn the channels' L/R switches off and assign those four channels to groups 1 and 2, creating a stereo submix. Then you would press the Group 1/2 To L/R switch, routing the submix to the main left/right bus. You could then control the overall level of

Studio 32 Specifications

Frequency Response	10 Hz–75 kHz, +0/-1 dB
Headroom	23 dB above nominal input
Maximum Gain	+76 dB; mic in to direct, group, and L/R outputs; balanced +70 dB, unbalanced +72 dB; mic in to monitor 1/2 out
Dynamic Range (typical)	111 dB (mic/line to main L/R output)
Signal-to-Noise Ratio	88 dB (mic/line to main L/R output)
Equivalent Input Noise (mic in to insert out)	-129 dB (at max. gain, 150Ω source)
Mic Output Noise	-91 dBu (min. gain, 150Ω source to insert)
Mix Output Noise	16 channels assigned: -83 dBu unbalanced, -77 dBu balanced; 1 channel assigned: -90 dBu unbalanced, -84 dBu balanced (faders nominal, trims minimum, 150Ω source terminations)
Distortion (THD+N)	at insert jack <0.001%; at main out (+21 dBu) <0.0015% (0 dBu signal into mic in, trim set for +15 dBu output at insert jack)
Weight	21 lbs.
Dimensions	19" (W) x 17.75" (H) x 6.6" (D)

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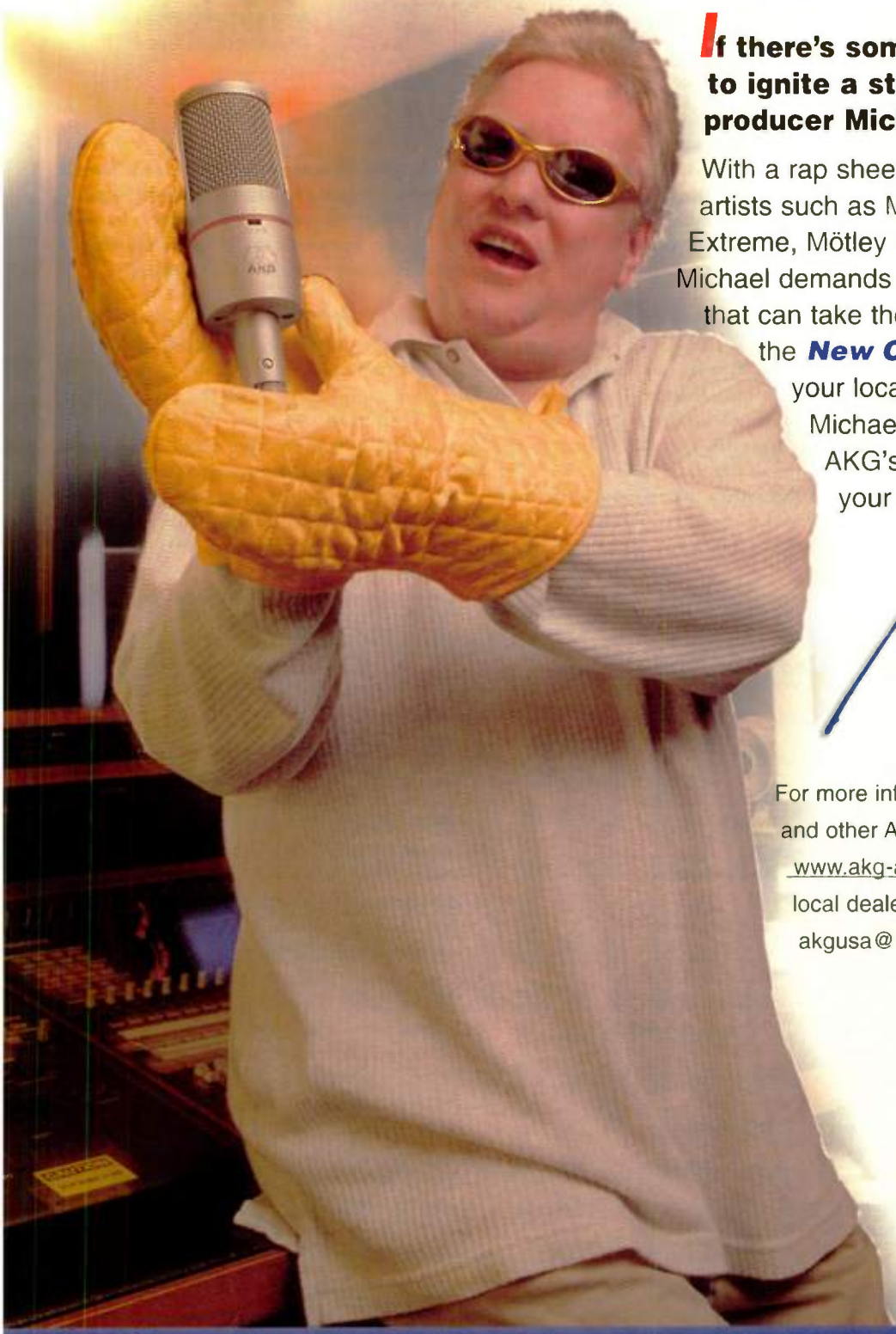

If there's someone who knows how to ignite a studio it's engineer/ producer Michael Wagener.

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the vocal submix with the group 1/2 faders. This technique is useful in live sound applications, as well as for complex studio mixes.

FEEDING THE CONTROL ROOM

You can switch the Studio 32's control-room monitor among several sources. With no buttons pressed, it defaults to the master L/R mix. By selecting the appropriate switch, you can listen to what's happening in monitor 1/2, aux 3/4 or 5/6, group 1/2 or 3/4, or the signal coming from the 2-track tape returns.

The solo master-level section has a level knob and a button to switch between Prefader Listen and Solo-In-Place, which is what we old-timers used to call "after-fader listen," or AFL. SIP is postfader and postpan, so it maintains the stereo separation of the soloed signals—and, therefore, your carefully crafted soundstage.

A Mono button sums the left and right sides so that you can check for phase problems and mono compatibility. Detecting phase problems is important, of course, but verifying mono compatibility is also a good idea, especially if you are preparing a recording that might get airplay. Many radio and TV stations still broadcast in mono or pseudo-stereo.

A pair of 10-LED ladders makes up the metering. These are peak meters, meaning that they respond to immediate level changes rather than the averaging you would get from VU metering.

ODDS 'N' ENDS

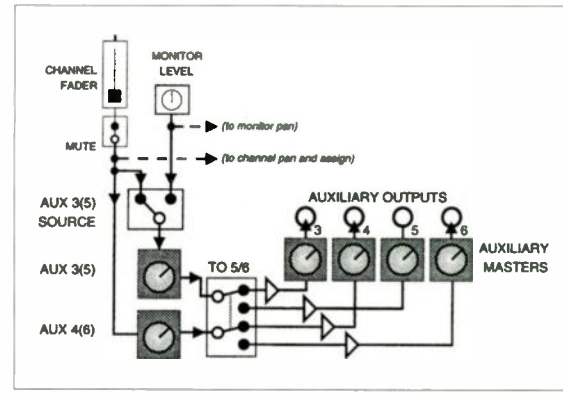
The Studio 32 comes with two jacks for headphone monitoring, with one level control. You can use headphones to monitor the signal that appears in the control-room output or the signal on the monitor 1/2 bus.

Rounding out the board's master section are LEDs that indicate the status of AC and phantom-power usage. The yellow phantom-power LED takes a long time (about 40 seconds) to go out after that power is turned off. Also, the phantom power is global—so if you turn it on, it's active on all the mic channels. It may just be a pet peeve of mine, but I have always disliked global phantom power; I prefer having it switchable on each input.

FIELD AND (AUDIO) STREAM

Now that the guided tour is over, let's start working with the board. The first thing we do when we get new gear is read the manual, right? Wrong! Okay, I admit it: When it comes to checking out new equipment, I just *have* to plug it in and start playing.

Eventually, though, I did actually read the Studio 32's manual (and no, I did *not* just look at the pretty block diagrams). I always approach this task with a degree of trepidation, because more often than not, these books are not only lacking in vital information but tend to be amusing in the most unintentional way. But much to my surprise, the Studio 32 manual presented me with useful information in an easy-to-read format. Homework done, I went right back to playing with the board.



COURTESY ALESIS CORPORATION

FIG. 2: The Aux 3(5) Source button determines whether auxes 3 and 5 carry the signal from the channel fader or from the monitor 1/2 bus. Either way, the sends are postfader. Two level knobs handle all four aux sends; the To 5/6 switch lets you choose whether the knobs control aux send 3/4 or 5/6.

Recently I agreed to do a live recording of the San Francisco Bay Area band Purgatory. I packed up the Studio 32 and a couple of ADAT XTs, and off I went. The gear all hooked up easily, and everything functioned the way it was supposed to. I tracked all the instruments and vocals using the direct outs on the back of the Studio 32 to feed the decks. I used the monitor 1/2 bus to listen to the ADAT tracks and used the aux outs to send a vocal feed to the house P.A. Everything went pretty smoothly, and for the most part I was happy. However, for this application, I

Studio 32 Features

Channel Inputs	(16) XLR mic; (16) balanced ¼" TRS line
Tape Inputs	(16) balanced ¼" TRS tape returns; (2) balanced ¼" TRS 2-track inputs
Direct Outputs	(16) balanced ¼" TRS
Inserts	(16) unbalanced ¼" TRS (tip = send, ring = return, sleeve = ground)
Aux and Monitor Sends	(4) balanced ¼" TRS aux; (2) balanced ¼" TRS monitor
Aux Returns	(8) unbalanced ¼" mono, configured as stereo pairs
Main L/R Outputs	(2) balanced ¼" TRS main L/R
Group Outputs	(4) balanced ¼" TRS
Other Audio Outputs	(2) balanced ¼" TRS control room; (2) stereo TRS headphone
Faders	(16) 100 mm channel; (4) 100 mm group; (1) 100 mm L/R master
Trim Range	mic input gain: 10–60 dB; line input gain: -10 dB (cut) to +40 dB
EQ	3-band: high shelf 12 kHz, ±15 dB; mid parametric 120 Hz–14 kHz, Q ⅛–2 octaves, ±15 dB; low shelf 80 Hz–1 kHz, ±15 dB
Low-Cut Filter	18 dB/octave below 75 Hz
Solo	solo-in-place, prefader listen
Level Meters	(2) 10-LED peak

would have liked to switch the mixer's direct outs to prefader; unfortunately, they can be only postfader.

Of course the big question is, How did the Studio 32 sound? The answer is simple (drum roll, please): It sounded great. The tracks came out clean, and the levels looked good.

A couple days later, the band came into the studio to fix mistakes and do some general sweetening. I configured the board in much the same way as for the live recording, with the playback coming through monitor 1/2, and I used the aux sends to set up a cue mix. We replaced some guitar and a few vocals. Because we were safely ensconced within the confines of a studio rather than working in an unknown environment, I used better microphones on the vocals. I am very familiar with these mics, and I wanted to know whether the Studio 32's preamps and other electronics changed their sound.

Upon listening to the playback, I found that the tracks sounded fine. The vocals came out crisp, clear, and uncolored. With four ADAT tracks left and three backup vocals still to do, I decided to use the Studio 32's groups to submix the backups to stereo. That left me with two ADAT tracks for recording the final mix. In addition, I wanted to find out whether there was any difference sonically when I took the feed from the group outs instead of from the direct outs. I didn't notice any difference, so we finished tracking and wrapped it.

The next day, I proceeded to mix the tracks. I set up a couple of reverbs, and I patched a compressor to an insert point and brought the reverbs back through the aux returns. I enjoyed the fact that I didn't have to repatch the cables from the deck; I just used the fader-source button. It all ran without a major hitch, and I was able to mix down with no complications.

I've already mentioned that this mixer is straightforward. But it is also versatile and user-friendly. You'll find a lot of different ways to patch things, and generally you can do your routing by simply flipping a switch. The Studio 32 even comes with convenient rack ears. Obviously, Alesis put a lot of thought into little design touches that simplify the music-production process. I found the Studio 32 reliable, easy to use, temptingly priced, and sonically clean. In my book, that's a great combination.

J.J. Jenkins *is an independent engineer, producer, musician, and bon vivant who lives on an island in the San Francisco Bay.*

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Studio 32 recording mixer
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FEATURES ■■■■
EASE OF USE ■■■■■
AUDIO QUALITY ■■■■
VALUE ■■■■
1 2 3 4 5

PROS: Good, clean sound. Separate mic and line inputs on all channels. EQ mid band is fully parametric. Tape returns are individually switchable between input channels and monitor 1/2 bus. Flexible routing of aux sends and returns. Easy to use.
CONS: Direct outs are postfader only. Phantom power is globally switched.
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STEINBERG

CUBASE VST/24 4.0 (MAC)

Bigger and better than ever, this high-end sequencer does high resolution.

By David Rubin

During the past few years, Steinberg's irrepressible *Cubase* sequencer has continued to grow by leaps and bounds. Like some nuclear-enhanced creature in a '50s sci-fi movie, this program just keeps getting more and more powerful. Fortunately, the mad scientists at Steinberg have harnessed this power for the benefit of the world's desktop musicians. High-end users, in particular, will appreciate many of the current version's enhancements. For example, *Cubase* now allows up to 96 tracks of 24-bit, 96 kHz digital audio (with the right hardware, of course). With its onboard mixer and ASIO bus system, *Cubase VST/24* supports a wide range of stereo and multi-I/O audio cards, including the Korg 1212, Lucid PCI-24, Event Layla, Sonorus StudI/O, Lexicon Studio, MOTU 2408, and Digidesign Project II.

Steinberg has also increased *Cubase*'s internal resolution to a surprising 15,360 ppqn for more accurate positioning and cutting of audio segments. Elsewhere in the program, you'll find a new Controller Editor window, a more powerful scoring section, a new MIDI mixer, and dozens of new tools and menu items. In fact, this new version of *Cubase* incorporates hundreds of subtle as well as distinct improvements and additions. Nevertheless, *Cubase VST/24* 4.0 looks and acts much like *Cubase VST* 3.5. Most of its changes are under the hood.

Cubase still has the same great-looking, functionally colorful, three-dimensional user interface; the same Arrange, Key Edit, Drum Edit, List Edit, Score Edit, Audio Editor, and Wave Editor windows; and the same architectural configuration of Parts, Tracks, Arrangements, and Songs. All the earlier version's features—such as the Logical Edit window and the Interactive Phrase Synthesizer—are still available, as are the powerful audio mixer and the hardware-rack-style effects. But the program has become more versatile and more sophisticated in many of its editing capabilities.

I did an extensive review of *Cubase VST* 3.5 in the November 1997 issue of **EM**, so from this point on, I'll avoid covering old ground and focus only on features that are new in *Cubase VST/24* 4.0. If you're unfamiliar with *Cubase*, you might want to read the earlier review (available on **EM**'s Web site at www.emusician.com).

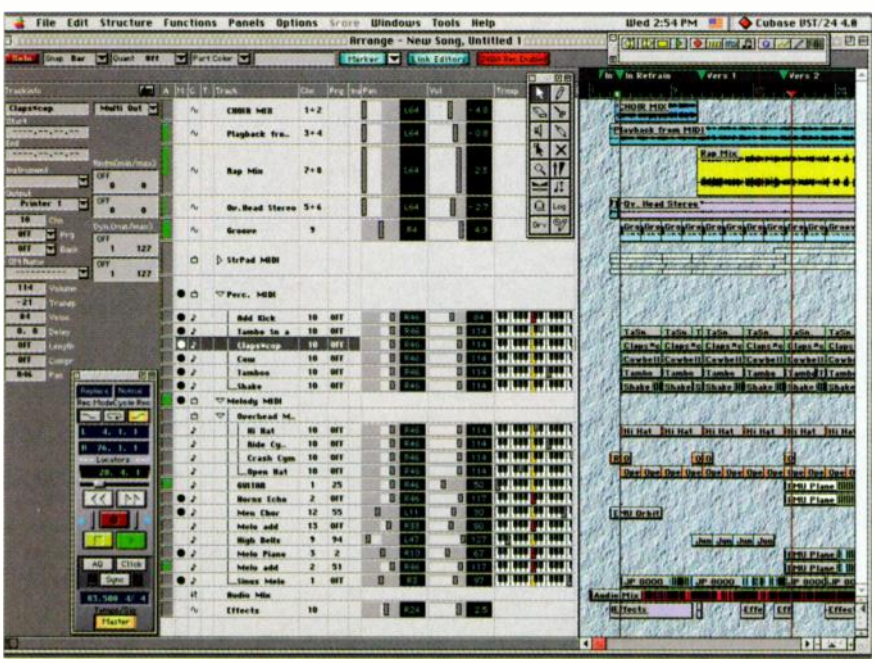

FIG. 1: *Cubase VST/24*'s newly enhanced Arrange window includes an expandable Inspector section with additional parameters. The Arrange window also lets you display faders for volume and panning and miniature keyboards for quickly transposing MIDI Tracks. The new customizable toolbar appears in the upper right, just above the new Markers Track.

MAKING ARRANGEMENTS

At first glance, the Arrange window (see **Fig. 1**) looks much the same as before, but start poking around, and you'll discover some great new enhancements. For example, the Inspector section can now be extended to reveal new parameter settings. The Randomize area lets you randomize note lengths, pitches, Velocities, and positions. The Dynamics area lets you limit or filter the Velocity values of any MIDI Track or Part without altering the original data.

The Arrange window now gives you much greater flexibility to customize the appearance of Parts and Tracks. As in the earlier version, you can adjust the width of columns by dragging the vertical dividers, but now you can also change the size of Tracks (individually or in groups) by dragging their horizontal dividers up or down. That's quite handy when, for instance, you have several waveform displays that you want to see more clearly, but you don't want to zoom in on the entire Arrange window. Speaking of zooming, the Arrange window's zoom controls (in the lower-right corner) now let you skip from one zoom level to another without cycling through them one at a time.

Steinberg has also added more graphic elements to the *Cubase* interface. You can now have onscreen sliders for panning and volume in the Arrange window. When you click on the Transpose field (in the Inspector), a miniature keyboard pops up, enabling you to change keys with a single mouse-click. In fact, you can now have transposition keyboards for each MIDI Track in the Arrange window. These graphic controls make an excellent supplement to the usual *Cubase* data-entry methods: scrolling values up or down with the mouse and entering numbers from the computer keyboard.

The computer keyboard, however, has hardly been neglected. *Cubase* now lets you assign your own key combinations (or MIDI notes) to virtually any menu item or editing tool in the program

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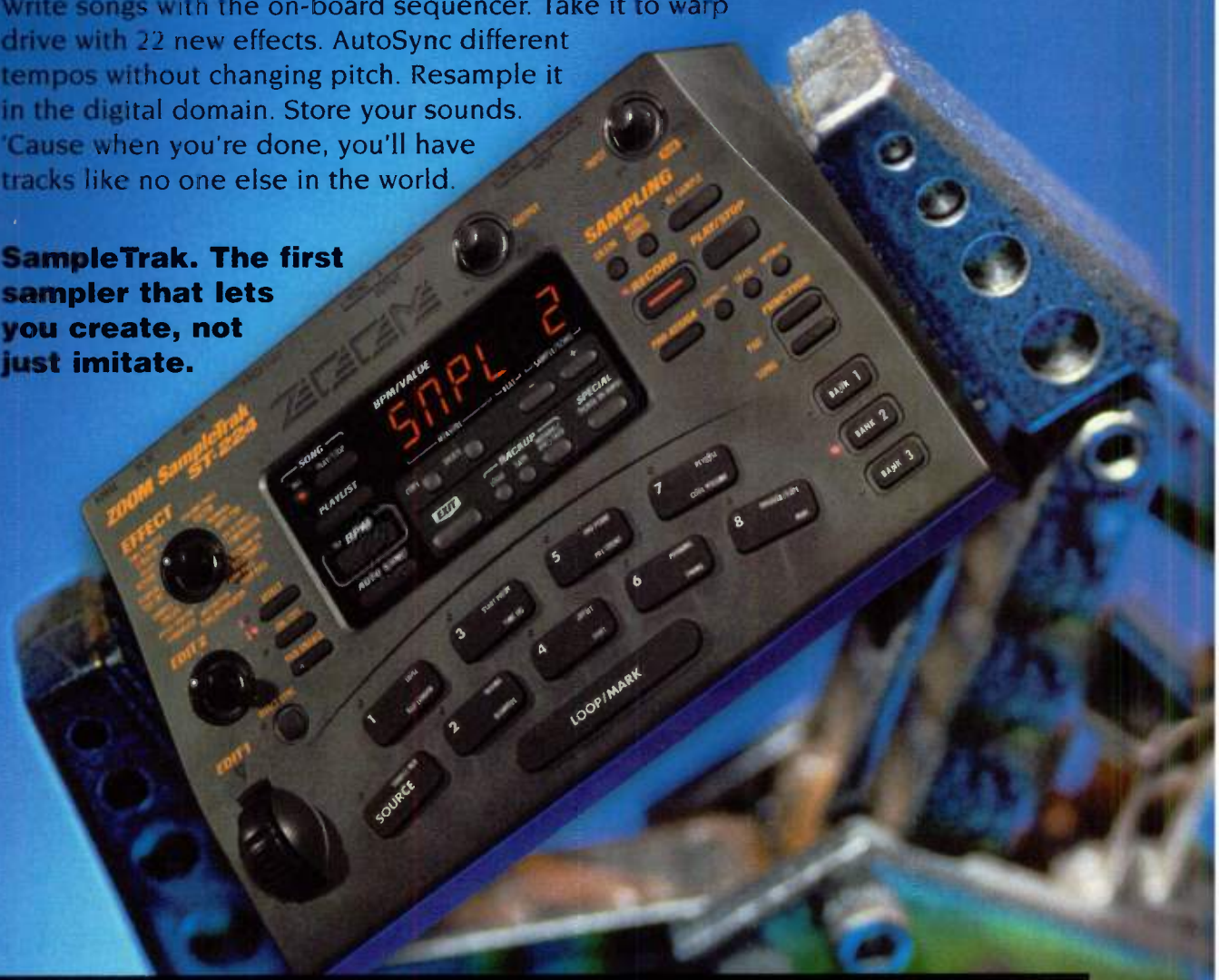

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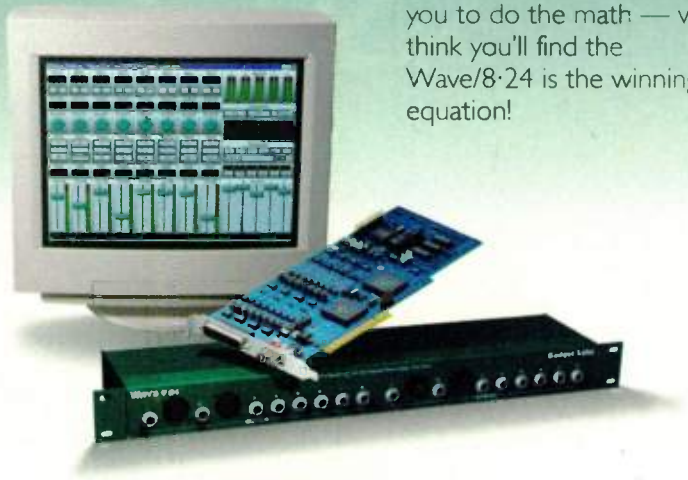

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(for both MIDI and audio Tracks). That should speed up repetitive operations quite a bit. Users who prefer buttons and icons are in luck, as well: the same menu items and editing tools can be added to and selected from the Arrange window's new customizable tool bar, so you can keep a collection of your favorite commands (including transport controls) within easy reach at all times.

Another significant enhancement to the Arrange window is the new Marker Track. You can now create and edit Markers just like other Parts. You use the Pencil tool to draw them in, and other tools let you copy, split, delete, name, and move them. A Marker pop-up menu lists the Markers by name with their start and end times. Double-clicking a Marker Part opens a text editor where you can keep detailed notes about each Marker.

As in the previous version of *Cubase*, you can view MIDI and audio Parts either as colored bars with labels or as miniature graphic displays. Version 4.0, however, lets you mix these view modes: you can have the audio Parts appear with graphic displays while the MIDI Parts appear with solid colors, or vice versa.

Furthermore, the Arrange window now includes special Folder Tracks. They let you combine several drum Tracks into a single percussion Folder, for example. Or you can group your brass Tracks and string Tracks in separate Folders to save space and help keep things organized. Unlike Group Tracks (which are still available), the Tracks in Folders (whether MIDI, Drum, or Audio) retain their separate identities and can easily be edited as needed. You can even keep Folders inside other Folders if your tastes run toward complex hierarchical structures.

THE CUBASE FAMILY TREE

In addition to its flagship sequencer, *Cubase VST/24* (which replaces *Cubase Audio XT*), Steinberg offers two less expensive versions of the program. *Cubase Score VST* ($549) is essentially the same as *Cubase VST/24*, except that it supports only 16-bit, 44.1 or 48 kHz audio and provides only 64 audio tracks. *Cubase VST* ($399) is now the low-end version of the program. It's the same as *Cubase Score VST*, except that it includes only basic scoring capabilities. If you're on a tight budget, and you aren't using high-end audio hardware, you may find these less expensive options to be attractive alternatives that provide the same *Cubase* user interface.

TOOLING AROUND

One of the most important improvements in *Cubase VST/24* is the addition of new tools and editing capabilities. For example, the program now supports full drag-and-drop editing of MIDI and audio Parts and MIDI events, and you can also drag and drop MIDI files to and from the desktop, the Arrange window, and various editing windows. That means, among other things, that you can back up an audio file by simply dragging it out of the Audio Pool window onto the desktop. You can also drag any combination of Parts or events into the Finder to create a new *Cubase* Part file, which you can later drag back into the program. MIDI and audio files can now be dragged directly into the Arrange window to add them to an Arrangement. Or you can drag them into the appropriate editing windows for close-up editing.

The Transport Bar has also been improved. It now offers five modes: large and small horizontal views and three vertical views (with various controls). The smallest vertical mode provides only the basic transport controls in a compact square. I especially like that configuration because I can tuck it into a corner where it's out of the way yet always available. A click of the mouse brings back the recording-option controls. In fact, if you add the transport commands to the new toolbar, you can do without the Transport Bar altogether in many cases.

Cubase VST/24's high resolution can become confusing when you're viewing meter position displays, marker locations, and event list entries. At a resolution of 15,360 ppqn, quickly determining the number of "ticks" in a dotted 16th-note, for example, will challenge the mathematical prowess of most users. To simplify the editing process, therefore, *Cubase* lets you choose a lower "display resolution" (7,680; 1,536; 960; 480; or 384 ppqn) so displays are easier to read and edits are easier to make. I prefer a screen resolution of 480 ppqn because I'm used to that number from working with other sequencers. But it really doesn't matter which setting you choose; *Cubase* always records and plays back at its maximum internal resolution.

As *Cubase* users know, each of the program's editing windows has its own

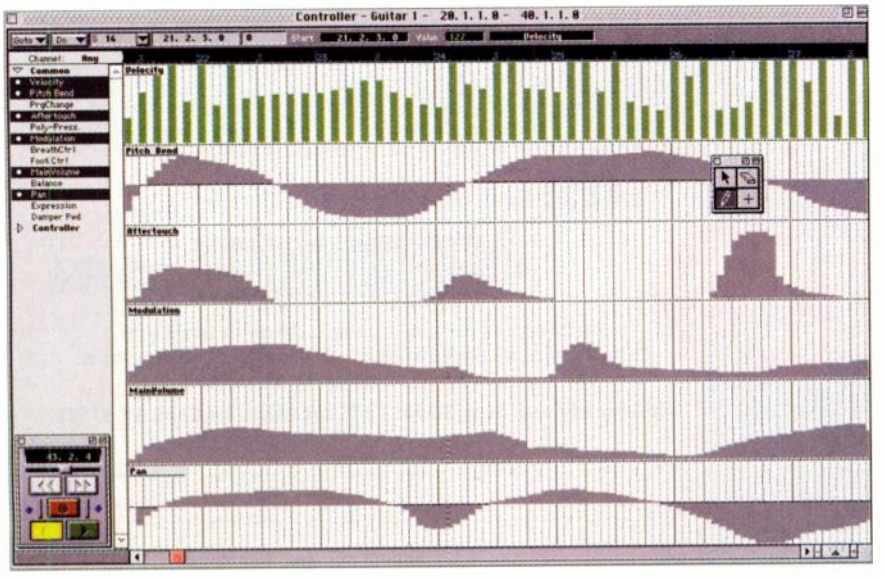

FIG. 2: *Cubase*'s new Controller Editor window lets you view and edit several types of Continuous Controller data at once. In the lower left, the Transport Bar appears in its smallest vertical format.

tools, which appear in a pop-up tool palette when you choose Tools from the menu bar. These Toolboxes have been expanded to include some valuable new editing tools. You can also assign the Toolboxes to key commands and tear them off the menu to position them anywhere onscreen, in vertical or horizontal orientation.

The new tools provide quick access to several important editing capabilities. For example, all Toolboxes now include a Speaker icon that lets you scrub individual MIDI notes and audition audio Parts in the Arrange window. In fact, you can use it to scrub MIDI notes in the Key Edit (piano roll), Drum Edit, List Edit, and Score Edit windows. (Audio scrubbing is available only in the Wave Editor window.) The Speaker tool is a great help for examining individual notes and for finding problem spots, but it only plays one note at a time. That means you can't play through chords and other groups of overlapping events in the Key Edit window, for example, to help you locate wayward notes. Even so, it's a handy tool.

The Key Edit, List Edit, and Drum Edit windows also provide a Mute tool, which lets you mute individual notes without altering the actual data. The Mute tool gives you a terrific way to quickly try out different combinations of notes without making the changes permanent. The same Mute tool is also available in the Arrange window, where it allows you to mute individual audio and MIDI Parts.

The Audio Editor's Toolbox sports a new Crossfade tool. It lets you crossfade between adjacent audio events, or you can use it to add fade-ins and fade-outs. Simply select the Crossfade tool and drag through the areas where you want the fade to occur. *Cubase* includes preset linear and logarithmic fade-in/out curves, but you can create your own curves by dragging over a graphic crossfade display with the mouse.

FIG. 3: *Cubase*'s groove-quantizing features have been improved and now include a dedicated Groove Box window for applying and editing grooves.

You'll notice a new icon in the Score Edit Toolbox, as well: the Cut Flag tool lets you replace a tied group of notes with a single long note. The Arrange window has the largest number of new tools, including dedicated MIDI tools for adjusting volume, panning, and transposition (with the pop-up keyboard).

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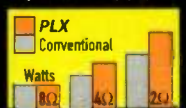

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FIG. 4: You'll need a pretty big monitor if you want to use all of *Cubase VST/24*'s audio and MIDI mixers and effects. The new MIDI Track Mixer appears in the lower left, showing several of the available XG and GS editing controls.

Other tools let you apply a groove directly to a Part, time-stretch MIDI and audio Parts, and apply Logical Edit presets (similar to search and replace functions) to Parts.

Perhaps the most powerful new tool in the Arrange window is the Selection Range tool. It lets you draw a selection box around any area in one or more Parts. The selection need not correspond to Part boundaries. In other words, you can select an area in the middle of one or several Parts and drag, drag-copy, delete, quantize, transpose, and otherwise edit the selected area as if it were a separate Part.

For example, you can now draw a selection box through the middle of Parts in several Tracks and copy and paste that musical fragment to create a recurring multi-instrument riff—something that wasn't possible in earlier versions. Clicking on a Marker Part selects the music on all Tracks within the Marker's range. Unfortunately, with all of its powerful new editing capabilities, *Cubase* still lacks a multiple-undo function. That would make a nice addition to the program.

WINDOWS GALORE

In its previous version, *Cubase* already had more editing windows than most people would use in a lifetime, but that hasn't stopped Steinberg from adding a few more. The new Controller Editor window is especially cool (see **Fig. 2**). It lets you view and edit several types of Continuous Controller data at once. Using a drop-down list, you specify which types of MIDI data (Pitch Bend, Aftertouch, Modulation, Panning, Breath Controller, and so on) to display. You can also view and edit non-continuous data types, such as Velocity, Damper Pedal, and Program Change. Making changes to the data displays is easy with the Pencil and Eraser tools.

The Controller Editor is not limited to MIDI Part messages. You can also use it to edit automation data created by the Audio Track and MIDI Track Mixers. (More on these a little later.)

Of course, *Cubase* still provides a controller display in the Key Edit and Drum Edit windows, and in most cases I prefer editing data there because I can correlate the controller data to the note positions. But if you use a lot of performance data in your sequences and you need to edit several types at once, the Controller Editor window should prove to be a powerful companion.

In another part of the program, Steinberg has substantially updated *Cubase*'s groove-quantize capabilities with the addition of a new Groove Box window (see **Fig. 3**). Grooves are now stored on your hard disk as separate files, treated just like Part files, and kept in a Grooves folder. The Groove Box displays the contents of the Grooves folder (which you can add to whenever you want) and lets you apply grooves to any selected Part or Parts. You can drag and drop grooves between the Groove Box and the Arrange window, and you can even audition a groove before committing to it. Timing, Velocity, and duration sliders let you change a groove's "grooviness." You can also edit grooves in the Key Edit and Drum Edit windows.

MIXED MESSAGES

Adhering to the adage "The more the merrier," *Cubase* continues to supply enough automated onscreen faders for you to mix yourself silly (see **Fig. 4**). In fact, this new version of *Cubase* can handle as many as 96 audio channels, if you have the appropriate hardware. Furthermore, each of those channels has eight effects sends to use with a new eight-space Channel effects "rack," which pops up when you click the FX button on the left side of the Audio Mixer window. The Master faders also have their own rack of four stereo Master effects, and you can add four separate Insert effects to each channel in the Mixer.

For those times when your mixes seem to be getting a bit out of control, *Cubase* includes a new Group Mixer. It lets you assign any combination of channels from the Audio Mixer to any of 16 automated Group channels. The Group channels have access to the same effects options as the Audio channels, so you can route any Group to four insert effects, eight channel effects, and four bands of EQ.

Other enhancements include a new Hi-Quality real-time EQ mode, which you can activate from the EQ window for each audio channel strip. (You can deactivate the Hi-Quality mode to go easy on your CPU.) I found that the Hi-Quality mode gave sax and guitar

Cubase VST/24 4.0
Minimum System Requirements
Power Mac 601/120 (604 or G3 CPU recommended); 32 MB RAM; Mac OS 7.6.1; Level 2 cache; 16-bit, 44.1 kHz built-in audio hardware or PCI audio card with ASIO driver; DMA hard-disk controller

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Jamie Rio
Gig Magazine, November 1998

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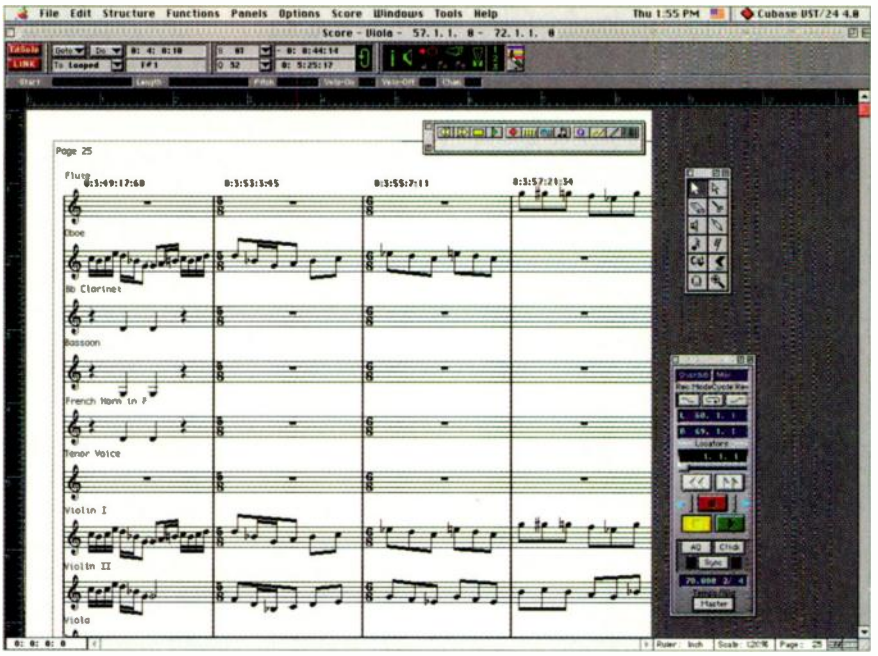

FIG. 5: The Score Edit window has been improved to include new tools and many new symbols and markings. In this example, the bar numbers have been replaced with SMPTE numbers.

Parts a noticeable boost in clarity and offered more precise control over the sound.

Cubase now offers a direct link to *ReBirth* 2.0, Steinberg's synthesizer and drum-machine emulation program. Through the company's new ReWire software link, you can stream *ReBirth*'s audio mix output or individual drum-sound outputs directly into the *Cubase* Audio Mixer, with sample-accurate sync. (Specialized channel strips automatically appear to accommodate the *ReBirth* outputs.) You can even control *ReBirth* parameters from within *Cubase*.

The MIDI side of mixing has also received some attention. *Cubase VST/24* now includes a MIDI Track Mixer window. It provides control over many of the same parameters that appear in the Inspector display. (It even supports Folder Tracks.) The colorful level meters indicate Note-On Velocities, while the faders control Volume. An Auto Configuration function automatically adjusts the Mixer to reflect the proper number of Tracks and shows the names of each. Parameter changes made in the Arrange window instantly appear in the Track Mixer and vice versa. Moreover, any channel in the Mixer can expand to include a set of controls for GS and XG instruments, so you can adjust and automate GS/XG effects and other parameters directly from the Mixer.

SCORING BIG

With version 4.0, *Cubase*'s Score Edit window (see **Fig. 5**) has evolved into such a full-featured notation section that it seriously rivals many dedicated notation programs. According to Steinberg, the current version of the program incorporates hundreds of changes, both large and small, which add up to a major overhaul.

Noteworthy improvements include several time-saving layout options. As in the previous version, instrument names are automatically added to the score from the Track names, but now you can copy brackets and braces from one group of staves and paste them onto other groups. In place of bar numbers, you can now display time-code numbers in the score, which should make film composers happy. And you can automatically add rehearsal markings based on your Markers. A new Auto Layout option also saves time when you're setting up such things as bar placement and staff spacing.

The current version of *Cubase* offers more flexibility in handling lyrics and chords than earlier versions did. You can now add lyrics by typing them into the Notepad (or another program) and simply pasting them into the score. *Cubase* automatically aligns the words with the notes. A "melisma" grab handle on the right of each syl-

lable lets you extend a line over any number of notes. A Search and Replace function makes finding and changing words easy and fast.

Chord symbols have become more customizable. You can select different fonts and text sizes for each part of the chord symbol, and you can create your own symbols and save them in a Chord Library. Guitar fretboard diagrams can now show as many as 12 frets, and Steinberg has added many new score symbols and markings. A Custom palette lets you assemble a collection of frequently used symbols for ready access, and a Words palette lets you create and save text symbols and performance directions. You can even import pictures into the score. Other enhancements make it easier to insert, move, and copy notes, symbols, and other score elements. And you can now add or remove tracks from view without leaving the Score Editor window—a great time-saver.

THIS AND THAT

Although *Cubase VST/24* comes on a CD-ROM, it uses a separate floppy disk as a key to authorize your hard drive. This copy-protection scheme has become

STEINBERG
Cubase VST/24 4.0 audio/MIDI sequencer
$799

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

PROS: Large assortment of editing windows and tools. Well-designed, customizable graphic interface with good use of color. Drag-and-drop editing. Powerful automated MIDI and Audio Mixers. Excellent effects-routing capabilities. Full-featured scoring section. Versatile Marker and Folder tracks.

CONS: Annoying copy-protection scheme. Full documentation available only in electronic form. No multiple-undo capability. Audio scrubbing available only in Wave Editor window.

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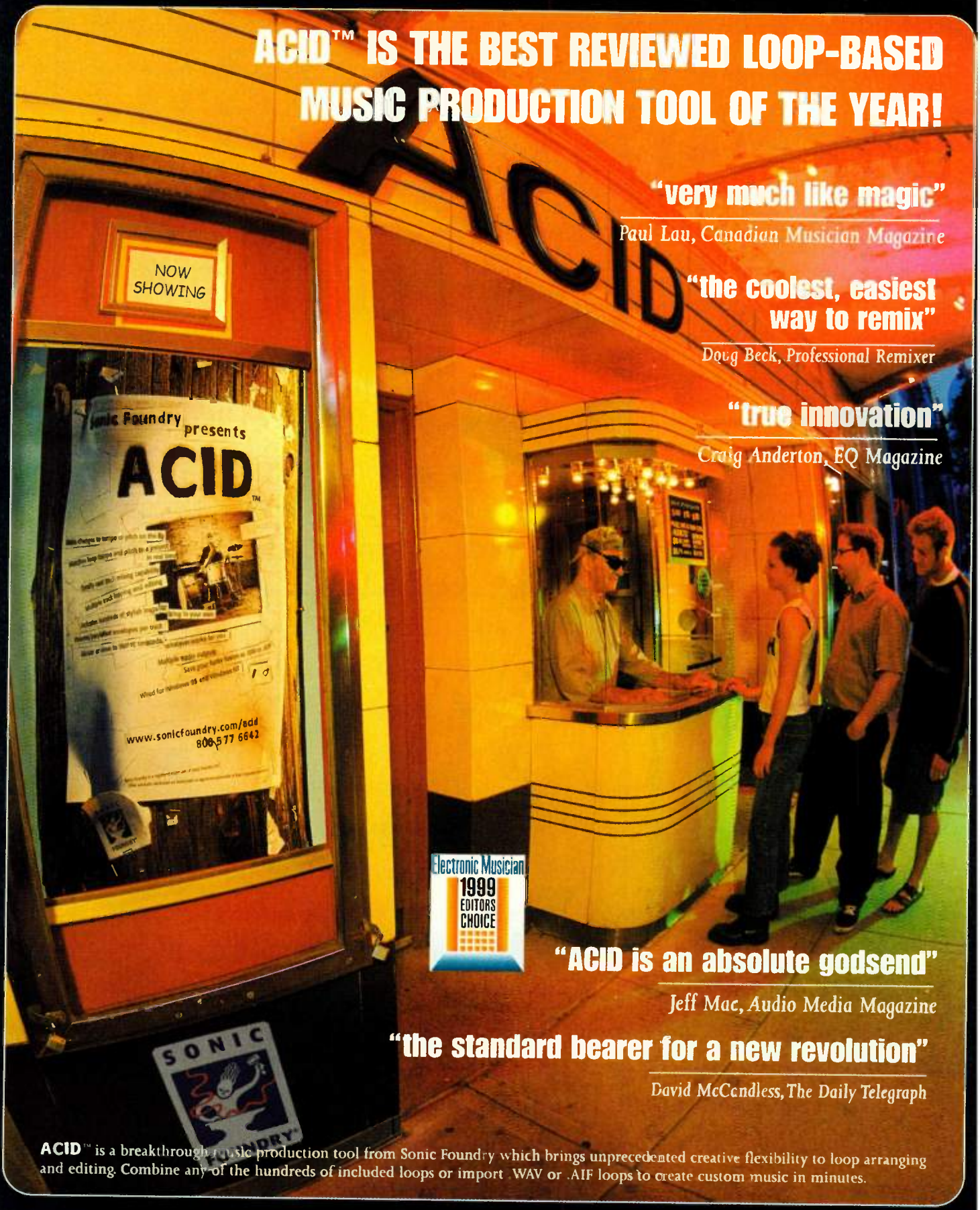

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quite common on the Macintosh platform, but Steinberg adds an extra twist: every so often, the program demands that you reinsert the CD for verification. That means you'll have to keep the CD close at hand or risk losing precious time hunting for it. I understand the need for copy protection, but this approach seems unnecessarily annoying. (According to Steinberg, by the time you read this, a different copy-protection scheme may be in place to accommodate new Macintosh models that lack floppy drives.)

The documentation that comes with *Cubase VST/24* 4.0 consists of a brief installation booklet and a poorly indexed 365-page *Getting Started* manual. The more "complete" documentation exists only in electronic form as a lengthy *Acrobat* file. This online manual includes color illustrations and links to related topics, but I still prefer to have a bound book in front of me while I'm using a program—especially a program as deep and complex as *Cubase*. Switching between programs and searching through online documentation quickly becomes tedious. Call me old-fashioned, but I still feel that, for an expensive, high-end program such as this, complete printed documentation should at least be available as an option. On the positive side, Steinberg includes a CD with an excellent QuickTime guided tour that highlights the main features of the program.

Copy-protection schemes and documentation shortcomings aside, however, the newest *Cubase* has much to admire. It has one of the best-looking user interfaces on the market and can be customized to a degree seldom seen in audio/MIDI sequencers. *Cubase* has so many kinds of tools and editing windows that it's hard to imagine not being able to tweak your data in whatever way you want.

For longtime *Cubase* users, the improvements in version 4.0 will likely feel organic and logical. The program works the way it always has, with a new level of versatility. *Cubase* now boasts even more graphic elements, more tools, more windows, and more options in general. Version 4.0 is an important evolutionary step for a program that just can't seem to contain itself.

Associate Editor **David Rubin** *lives and works in the suburbs of Los Angeles.*

EARTHWORKS

Z30X

A cardioid condenser microphone with a ruler-flat response.

By Rob Shrock

If the name Earthworks doesn't immediately ring a bell for you, that's understandable. In the past, the company has devoted most of its time to manufacturing high-quality omnidirectional microphones, such as the QTC1 and TC30K, and super-accurate measurement microphones, like the M30 and M55. All these mics are exceptional, but they are not well known among personal-studio owners. Let's face it, for the average recordist on a budget, directional mics are the name of the game.

With the Z30X, Earthworks has produced a mic that will appeal to a much broader audience. The Z30X is a fixed-pattern cardioid condenser mic designed for studio applications. It shares the unique look of previous Earthworks microphones, sporting a pencil-style design with a tapered front end. You can purchase two of these mics as a matched pair, complete with an elegant cherry wood carrying case. A hypercardioid version, the Z30XL, is also available (see the sidebar "More Earthworks Offerings").

HEART OF GOLD

The Z30X's maximum rear rejection is at 180 degrees; in other words, the mic captures sound mainly from in front of the capsule and rejects sound from behind. (The area covered by the mic is heart shaped, hence the name *cardioid*.) The Z30X's tapered design makes it easy to set up: just point it at your source as you would point a flashlight, and it will be directly on-axis.

The frequency response of the Z30X is very smooth, so you can minimize the proximity effect by moving the mic back a bit without losing crucial midrange frequencies. The result is a realistic and natural sound that is similar to what you'd expect to receive from an omnidirectional microphone. Even when I shoved it close to a source for an intentionally bass-heavy effect, the Z30X retained its punchiness and top-end clarity.

In addition to controlling the bass-frequency content by adjusting the miking distance, you can attenuate high frequencies by turning the Z30X slightly off-axis. I found the Z30X to be most neutral sounding when pointed on-axis at a source about 12 to 14 inches away.

DEAD MAN'S CURVE

The Earthworks design philosophy is to produce mics with ruler-flat frequency responses. The result is a line of natural-sounding mics that don't impart coloration to the signal. The Z30X is no exception, delivering a wide frequency response (30 Hz to 30 kHz) with minimal deviation from 0 dB (typically ±1 dB).

To that end, the Z30X does not exhibit the upper-midrange frequency boost commonly associated with many popular cardioid condensers, so it might not "grab" you on first impression. In fact, the Z30X is so revealing that it's almost disheartening: many flaws of the recording environment (mic placement, instrument tonality, and so on) become painfully obvious when you use this mic. On the other hand, when you absolutely love what you are hearing from an instrument or voice, the Z30X could be the perfect mic for getting that sound into your recorder.

Adjusting to the wide, flat response of the Z30X took me a bit of time because I'm used to grabbing certain microphones for those applications in which I know the tonal response of the microphone will complement the source. However, I soon grew to love the natural response of this mic, especially when used in a stereo combination. Capturing an instrument with the frequency balance intact allows you to add or subtract frequencies at the console without fighting against an already colored signal. In fact, when recording with the Z30X, I found that I needed less EQ overall.

Tracks recorded with the Z30X pair also exhibited some notable qualities on playback. I have recently been monitoring on Genelec, Event, Mackie, and Audix monitors, and, although each monitor sounds different, the Z30X tracks sounded full and natural on all of them—most of the time with minimal EQ. I am convinced this is due to the balanced frequency response of the mic.

The Z30X tracks also sounded great when I listened from locations other than in front of the speakers. One test I frequently use is to listen to how the track sounds when you're down the hall

For manufacturer information, please see Contact Sheet, p. 163.

The Z30X is the first cardioid microphone from Earthworks, a company best known for omnidirectional mics. Featuring a flat frequency response and high SPL rating, the Z30X is a versatile mic that can accurately capture a sound source without adding coloration.

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or in the next room. If the sound is good only when your face is jammed in front of a pair of $1,500 speakers, something is wrong. I'm happy to report that the Z30X passed this test with honors, too.

PIANO MAN

Grand pianos cover almost the entire frequency spectrum, so I have always considered recording them as a splendid way to test condenser mics. My first test involved using a pair of Z30Xs to mic a solo piano for a corporate multimedia score.

For this session, I recorded four stereo pairs to an ADAT XT20: the Z30Xs with an Earthworks LAB102 preamp, which is a very transparent unit (see the review in the February 1999 issue of **EM**); a pair of AKG 451s through API preamps; a pair of Neumann U 87s through API preamps; and a ceiling-mounted pair of Crown PZM room mics run directly into the SSL console. Because this project was on a tight deadline, I felt the chosen mic array would guarantee an acceptable recording and also give me a framework for evaluating the Z30Xs in action.

The pairs of Z30Xs and 451s were placed in XY configurations just behind the hammers and about 18 inches above the soundboard. I positioned the two pairs as close together as physically possible (without interfering with each pair's pickup patterns), which allowed me to later make a direct comparison between the Z30Xs and a fairly common combination of mic and preamp often used for recording piano. The U 87s were set up about 12 feet back to capture some of the ambience of the nice room we were working in, which was designed by Russ Berger.

Initially, the Z30X pair failed to grab my attention, especially when compared with the 451s. The 451/API combination exhibited more bite in the upper mids (2 to 3 kHz) and a bump in the lower mids (around 200 Hz)—a bit of an overall "hyped" effect. Although this sound is immediately recognizable—and often

Z30X Specifications

Frequency Response (@15 cm)	30 Hz–30 kHz (±1.5 dB, typically ±1 dB)
Polar Pattern	cardioid
Sensitivity	10 mV/Pa (-40 dBV/Pa)
Power Requirements	48V phantom power, 10 mA
Maximum Input	145 dB SPL with 5 kΩ load
Peak Output Voltage	3V into 1 kΩ, 10V into 5 kΩ
Output	XLR balanced
Minimum Output Load	600Ω
Noise	22 dB SPL equivalent (A weighted)
Dimensions	9.25" (L) x 0.86" (D)
Weight	0.5 lb.

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MORE EARTHWORKS OFFERINGS

The Z30XL is identical to the Z30X in specifications, with the exception of its polar pattern. The Z30XL's hypercardioid pattern features a 130-degree maximum rear rejection, making the Z30XL more directional than the Z30X. Because less bleed is picked up by the Z30XL, it is better suited for situations that call for maximum isolation. This greater isolation comes at a price—the Z30XL lists for $150 more than the Z30X ($900; matched pair with case, $1,900).

Earthworks also recently released the SR71, which is basically a version of the Z30X designed for live-sound applications. Housed in a satin-black brass body, the SR71 is more durable than the Z30X and Z30XL and can withstand drops, knocks, and drumsticks better. The frequency response is rated at 50 Hz to 20 kHz (±2 dB), which is less spectacular than its big siblings, but if the SR71 is anywhere near as good as the Z30X, you can't go wrong—especially for $399.

desired—as a pop piano timbre, it did not accurately represent the true sound of the piano. The transparent Earthworks signal path sounded somewhat stark by comparison, but after a few minutes, I realized that it was similar to what I was hearing from the piano when standing in the room with it. Because I was not completely at home with this particular studio's selection of monitors, we made no adjustments other than some tweaks in the mic placement.

Back at my studio, I was able to sort through the tracks, find the mic combinations I liked best, and learn how well the tracks recorded with the Z30Xs translated to my monitors. With a few EQ adustments, I was also able to simulate the 451/API sound using the Z30X tracks (although that was not the sound I was looking for on this particular project). Because the piano score was intended to reside under dialog, I ultimately settled on a combination of the unequalized Z30X and ceiling-mounted PZM tracks. I really liked the flexibility the Z30Xs gave me at mixdown.

GREASED LIGHTNING

Encouraged by the Z30X's performance on the solo-piano date, I was eager to test the mic on other instruments. Another positive characteristic of Earthworks microphones is their excellent transient response, due largely to the small capsules used in their designs. (The larger the capsule, the more air mass needs to be moved to register an impulse.) There is a common belief that small-capsule mics are not ideal for capturing low frequencies; but I've found that mics with small capsules can record both bass frequencies and percussive transients equally well. The Z30X supports this premise: its small capsule captures frequencies well below the frequency responses of most microphones. There are times when a large-diaphragm mic is the preferred choice, but not as a standard default.

To check this out further, I used the Z30X to record the instrument that is the most difficult to get right on a consistent basis: the human voice. I recorded a male and a female soloist using the Z30X routed through a custom Audio Upgrades preamp and an EL-8 Distressor, and I loved the results. Once again, the mic produced a full-bodied, natural sound. Although I did add EQ to the vocal tracks during mixdown, it was just to sit the track better in the

EARTHWORKS
Z30X cardioid condenser microphone
$750 each;
matched pair with case $1,600

AUDIO QUALITY ■■■■■
VALUE ■■■■■
1 2 3 4 5

PROS: Ruler-flat frequency response from 30 Hz to 30 kHz. Natural sound. Accurately captures fine details. Can handle high SPL. Low handling noise. Translates well across a variety of monitoring systems.
CONS: No pad or low-frequency rolloff. Fixed pattern.
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† Price not including the cost of a PC with audio (approx $1,350) PII/300Mhz, 128 MB RAM, 8.4GB UDMA, 36X CD ROM, sound card w/ S/PDIF, 4MB SGRAM Graphics, FD, Win98, kbd, mouse and monitor.

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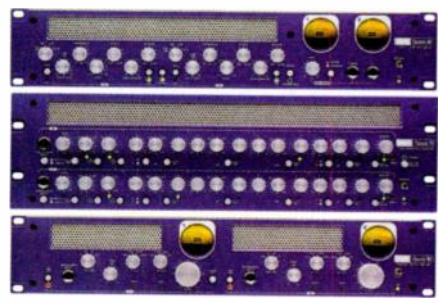

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mix, not to compensate for deficiencies. I did employ a highpass filter to get rid of unwanted frequencies and room noise below 70 Hz, which the Z30X picked up all too well. Despite this, though, the Z30X has very low handling noise—so low that a singer could hold the mic during recording.

I also recorded acoustic guitar and glockenspiel with the Z30X, with excellent results. The glockenspiel (like triangle, tambourine, and finger cymbals) is particularly difficult to record accurately because its high-frequency overtones are quite complex and extend well above the 20 kHz limit commonly found in digital recording devices. I recorded the glock into an Ensoniq PARIS system at 48 kHz, with 24-bit resolution; although it sounded much better than it would at 44.1 kHz with 16-bit resolution, the system still couldn't record all the high frequencies that the Z30X is capable of delivering. The transients from a glockenspiel can also be difficult to reproduce, but the Z30X responded very well by preserving the immediacy of the mallet hitting the metal. I would love to hear the Z30X when 24-bit/96 kHz converters are the norm for digital recording.

UP, UP, AND AWAY

Once again, Earthworks has created a solid winner. The Z30X delivers the flat, extended response we've come to expect from Earthworks mics. There are a few minor drawbacks, however. For instance, the Z30X does not offer a pad, although it can handle up to 145 dB of SPL, so you probably won't need one anyway. Also, it has no bass rolloff, so you are going to get all those very low frequencies at the mic whether you want them or not—it's up to you to filter them later. But these are not major problems.

If you're looking for coloration from a microphone, look elsewhere. The Z30X is designed to accurately capture the surrounding sound waves without adding any extra flavors to the recipe. However, if you use cardioid mics for intentional proximity effect or tonal control, the Z30X will also serve you well by providing extended low and high frequencies with excellent transient response. This mic is great for just about anything: vocals, guitar, percussion, pianos, you name it. A pair of Z30Xs would make a great addition to anyone's microphone arsenal. ❁

ACCESS

VIRUS

A compact and versatile analog synth emulator.

By Craig Negoescu

FIG. 1: The Virus's six unbalanced ¼-inch outputs can be configured as three pairs of stereo outputs. One of the six doubles as a stereo headphone jack. The audio input jacks feed into the filter section and allow you to apply distortion, panning, amplification, envelopes, and effects.

Germany's Access Music Electronics built its reputation by making hardware front ends for the Oberheim Matrix-1000 and the Waldorf MicroWave. Recently, Access has put its expertise into producing the Virus, a compact, yet versatile, physical-modeling synthesizer designed to emulate analog synths.

The Virus is about the size of a computer keyboard, making it small enough to sit comfortably on your desktop. Its powerful synth engine operates on a single Motorola 56303 chip, which is well suited to the demands of real-time synthesis. The Virus offers 12-voice polyphony that can be dynamically shared across 16 multitimbral parts, with enough power left over for external audio processing and real-time effects.

MAKING CONNECTIONS

On the rear panel (see **Fig. 1**) are six unbalanced ¼-inch outputs, configurable as three stereo pairs. One of the six audio outputs doubles as a stereo headphone jack. A pair of audio input jacks are provided to bring the world into the Virus for treatment. The input signal feeds into the filter section and takes advantage of distortion, panning, amplification, envelopes, and effects.

The unit has MIDI In and Out ports but no MIDI Thru port. An internally switched MIDI Soft Thru allows you to route messages received at the MIDI In jack to the MIDI Out.

The front panel sports 32 knobs, 27 buttons, and a 2 × 16–character LCD. The tempo LEDs for LFOs 1 and 2 can be reset to display input clipping. The knobs have a slightly rubbery feel, with molded gray markers, and the buttons exhibit a nice, tactile click. Rubber feet on the bottom of the unit help prevent it from slipping. Overall, the unit feels solid and roadworthy. The Virus is powered by a lump-in-the-line external power supply, and it can be rackmounted with optional adapters ($29).

FACTORY FRESH

The Virus ships with 256 Single patches and 128 Multis (combinations of Single patches). All these patches can be edited; many of the Multi patches split the keyboard between arpeggiations in the lower half and a lead sound in the upper half. The more complex Multis layer their sounds across different MIDI channels.

One complaint I have about the factory Multi patches is that they favor techno and disco sounds over vintage, ambient, and grunge sounds. Additionally, the rhythm-based Multi patches tend to be a bit monotonous because of the arpeggiator's lack of features.

However, the 256 Single presets, which are divided between two banks, show the incredible range of features that the Virus has to offer. Bank A consists of percussive synth basses, analog- and PPG-like pads, filtery arpeggiators, and a variety of synth-percussion sounds. Bank B contains a similar range of sounds but also includes patches for modifying external audio inputs.

AS EASY AS YOU LIKE

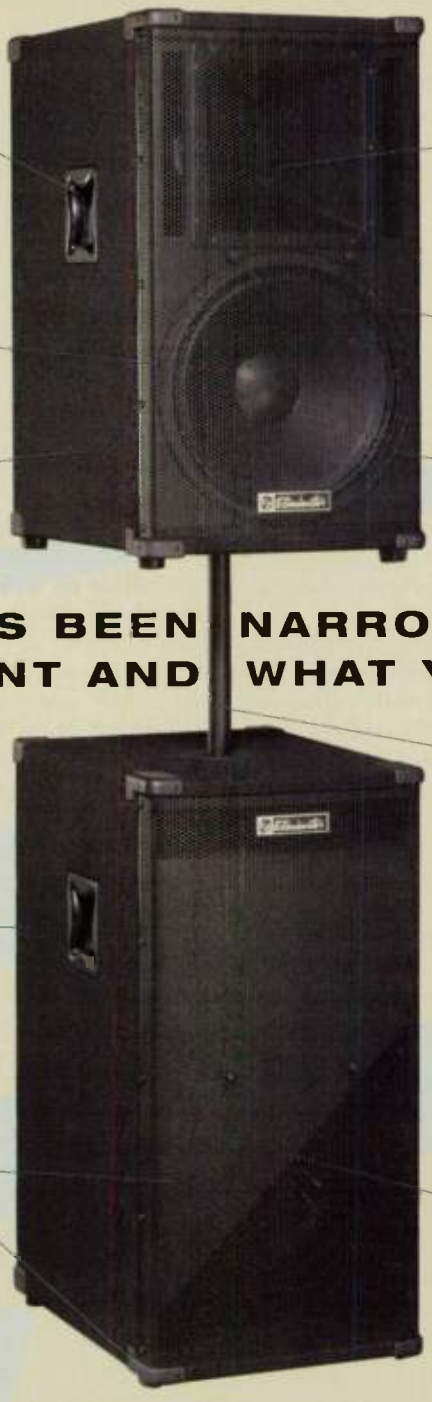
The Virus has two editing modes: Easy and Expert. Easy mode gives you access to only the basic menu parameters, while Expert mode gives you full control of all parameters and delves deeper into menu arcana.

The editing process is straightforward, making effective use of the limited display size, and includes a number of helpful shortcuts. Holding one parameter button down while hitting another jumps you from one edit category to the next; you can then use the increment and decrement buttons to home in on your target. You can hold down both value buttons at once to jump to zero.

You can also define how the display responds to knob movement. Normally, when a knob is turned, the display changes in response. If you wish, you can set the display to snap back after changes are made, linger for a while on the parameter, or ignore knob movement altogether. Similarly, you can configure the knobs to change values relatively or absolutely, or you can turn the knobs completely off for safety.

The Virus manual is thorough and well organized. The opening chapter functions as a primer, with subsequent chapters going into greater detail. Software upgrades and manual updates are available online from the Access Web site.

The Access Virus is a compact physical-modeling synthesizer designed to emulate analog synth sounds. Features include an arpeggiator, vocoder, and easily upgradable operating system.



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EV Eliminator

Updating the Virus operating system is easy. While I was writing this review, several new features were introduced. All I had to do was download the upgrade from the Access Web site, drop it into a MIDI sequencer, and send it to the synthesizer. After a little waiting and a few menu prompts, my review unit had the new operating system in its flash memory. (According to Access, by the time you read this review, a new update will be available that will include a modulation matrix and a ring modulator.)

MAKING WAVES

The voice architecture of the Virus consists primarily of four sound sources: two main oscillators (which can be synched), a suboscillator (tuned an octave below oscillator 1), and a noise generator. From the front panel, you can choose between a sawtooth, the wave generator, and a pulse wave (with an adjustable pulse width). The wave generator offers a sine wave; a triangle wave that can be reshaped into a sawtooth or square wave by using the Shape knob; and 62 additional single-cycle waveforms, bringing the total number of waves to 66. The wave generator's waveforms are accessed under the Edit menu.

When you're ready to fatten up your sound, dial in a little suboscillator and choose a square or triangle wave in the Edit menu. From the Edit menu, you can also access Twin mode, which adds a double of the voice being played. Twin mode allows you to pan each of the voices separately in the stereo field, as well as shift the phase of each voice's LFOs. The one drawback of using Twin mode, however, is that you sacrifice some of your polyphony.

FILTER THIS

The Virus features two filters that can be independently set to lowpass, highpass, bandpass, or band reject. The filters share front-panel knobs for Resonance, Envelope Amount, and Keyfollow, but each filter has its own dedicated Cutoff knob. In addition, the filter section has a 5-stage envelope generator that can be applied independently to each filter in a positive or negative direction.

A button on the front panel allows you to switch between four filter configurations (see **Fig. 2**), the first of which is a serial configuration that creates a 4-pole filter. The second configuration stacks a 4-pole and a 2-pole filter to create a silky-smooth 6-pole filter. The third configuration arranges the filters in parallel.

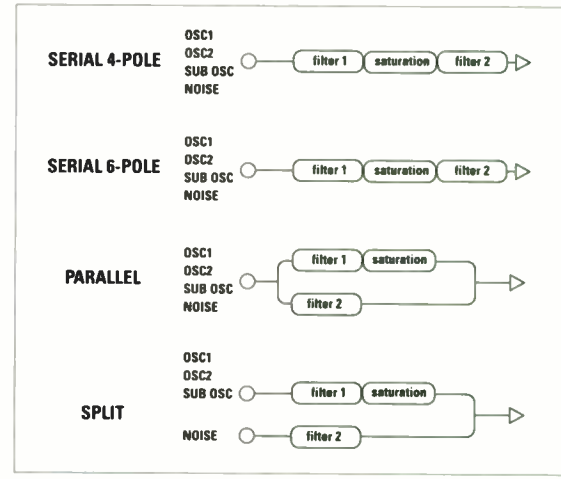

FIG. 2: The Virus offers four filter configurations. The fourth one allows you to pan each of the filter outputs separately to fatten up your sound.

The fourth configuration is unique in that it sends all oscillators to filter 1, leaving the noise generator for filter 2. When the filters are split like this, you can vary the stereo spread of their outputs for a "stacked synth" effect, using only one voice.

As a nod to the grunge-synth style of the '90s, Access thoughtfully included a saturation stage after filter 1. A number of saturation curves are included, ranging from gentle to digital distortion, allowing you to add some serious grit to your sound.

VOCODER DELIGHTS

With Virus OS 2.0, the filters can be configured to emulate a classic analog vocoder. The vocoder uses quite a bit of DSP power, however, requiring up to four voices in 32-band operation. A menu is provided that allows you to select from 1 to 32 bands for the vocoder effect, and the inputs can be internal, external, or both.

When the vocoder is active, the conventional filter architecture is disabled, and the filter-section knobs are automatically assigned to vocoder parameters: the Filter Cutoff adjusts the vocoder's center frequency; Keyfollow controls the bandwidth of the carrier or modulator, based on the Filter Select buttons; and the Attack and Decay knobs adjust the vocoder envelope. Additionally, carrier and modulator spectra may be shifted against one another for exotic spectral-inversion effects. This reassignment of the filter controls affects only the Multi part where the vocoder is active.

Virus Specifications

Synthesis Engine	physical modeling
Maximum Polyphony	12 voices
Multitimbral Parts	16
Presets (RAM)	384
Waveforms	66
Filters	2
Envelope Generators	2
Number of LFOs	3
LFO Waveforms	5: triangle, sawtooth, square, sample-and-hold, sample-and-glide
Arpeggiators	16
Number of Effects	(4) chorus; (1) delay
Inputs	(2) ¼" unbalanced
Audio Outputs	(6) ¼" unbalanced
Other Ports	MIDI In and Out
Display	2-line x 16-character LCD
Dimensions	18.35" (W) x 2.36" (H) x 7.09" (D)
Weight	5.5 lbs.

For manufacturer information, please see Contact Sheet, p. 163.

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ENVELOPE, PLEASE

Both the filter and amplifier are controlled by 5-stage (ADSTR) envelopes. The added Time control, located after the sustain portion of the envelope, allows you to determine when the sustain phase of the envelope will end. With the Time knob in the 12 o'clock position, the sustain portion behaves as in a conventional 4-stage ADSR envelope. Set the Time knob to the left, and the sustain is shortened; move it to the right, and sustain time is increased.

▼

The editing process makes effective use of the limited display size.

The Envelope Edit menu also includes a Punch parameter that allows you to increase the intensity of the envelope attack. You'll want to use Punch when creating aggressive drum and bass sounds.

LFO MY!

The parameters of LFOs 1 and 2 are accessible from the front panel, while LFO 3's parameters are available only in the Edit menu. All three LFOs can be internally or externally clocked and come in a variety of wave shapes: triangle, sawtooth, square, sample-and-hold, and sample-and-glide (similar to sample-and-hold but with portamento).

The LFO triangle wave can be morphed into a positive or negative sawtooth shape. I was mildly disappointed, though, to find that the square-wave duty cycle and the sample-and-glide lag time cannot be adjusted. One nice feature is that LFOs 1 and 2 have a Start-Phase control that can be set to begin modulation at any point in the LFO cycle.

Although the three are functionally identical, each LFO has unique assignments. LFO 1 can be assigned to control oscillator frequency and pulse width, to overall filter gain, and to the resonance of both filters. LFO 2 can be used to control waveshaping, FM amount, the cutoff of filters 1 and 2, and panning. LFO 3 is a junior version of LFO 1 but with an added fade-in

control for creating natural-sounding vibrato.

LFOs 1 and 2 can also be reconfigured into simple (but useful) attack-decay envelopes, with rise and fall times set by the rate knob. When you include the envelopes for the filter and amplifier, you end up with a total of four envelope generators on the Virus.

BUT WAIT! THERE'S MORE!

The Virus features two basic internal effects: chorus and delay. Chorus can be assigned to only 4 of the 16 Multi parts, but delay is applied globally. Both effects have rate, shape, depth, and feedback parameters.

Although the Virus includes an onboard arpeggiator, a number of essential features have been omitted, such as keymapped patterns and TB-303–style step sequencing. The arpeggiator can sync to MIDI Clock at various clock divisions, and the patterns can be directed up or down or randomized. Considering the extraordinary sonic depth of the Virus, it's unfortunate that a more sophisticated arpeggiator wasn't included.

Because much of the Virus's editing takes place deep within menus, Access included two extra front-panel knobs—labeled Definable 1 and 2—that give you a handy place to tweak your favorite synth or MIDI parameters. In many of the presets, these knobs are already mapped to control effects or arpeggiator tempo.

VIRUS FEVER

The Virus has everything an analog-synth lover could want in a digital model: a wide variety of oscillator waveforms, sophisticated filtering, versatile envelopes and LFOs, external audio inputs, a vocoder, and numerous routing options. The patches emulate classic analog-synth favorites, as well as contemporary hybrids, and they sound great.

In a feature-to-feature comparison, the Virus rates above much of the competition, making this instrument an ideal choice for anyone interested in a powerful but compact analog-modeling synthesizer.

Craig Negoescu *has spent the last 23 years turning knobs and pushing buttons. Currently, he's helping change the world over at frogdesign (www.frogdesign.com).*

ACCESS
Virus analog-modeling synthesizer
$1,695

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

PROS: Compact design. Six discrete audio outputs. Oscillator wavetables greatly expand the sonic palette. Flexible filter routing. Filter saturation stage. LFOs can double as EGs. Parameters change in real time without stepping noise. Interface flexibility for programming or performance use. Software upgradeable OS. Two audio inputs. 32-band vocoder.

CONS: Arpeggiators are limited. No LFO self-triggering. Effects section lacks sophistication.

CIRCLE #442 ON READER SERVICE CARD

ARBORETUM

HYPERPRISM (MAC/WIN)

Plug-in power to jump-start your audio.

By Dennis Miller

Arboretum's *Hyperprism* real-time effects plug-ins have gained nearly cult status in the desktop music world. Due in part to their unique user interfaces and high-quality processing routines, the software has become a standard in many high-end music circles. With the release of version 2, for both Mac and Windows, *Hyperprism* should become even more widely known in the audio community. Sporting new and optimized effects and DirectX support, *Hyperprism* is a powerful addition to any audio arsenal.

Hyperprism is a set of several dozen effects that can perform numerous sound-altering feats. These range from fairly standard processes, such as reverb and delay, to less common functions, such as the Phase Vocoder and Z-morph (Mac only). The program offers a consistent look and feel for all the effects and gives you powerful real-time control over their various settings.

Hyperprism is available in several versions, each with slightly different configurations and requirements. (See the sidebar "Hyperprism Versions" for a list of the different formats and their current release numbers.) Premiere, VST, AudioSuite, and TDM plug-in versions are available on the Mac, and there's also a dedicated version for use with the included *HyperEngine* host program. The *HyperEngine* version offers significant enhancements over the plug-in versions, including several new effects and a highly optimized interface. On the PC, the only option is the *Hyperprism-DX* plug-in pack, which is intended for use with a DirectX host. This version offers features similar to those of the Mac plug-in versions. I'll cover the main features of the *Hyperprism* plug-ins that apply to both the Mac and PC in the main section of this review; you can read about the *HyperEngine* version in the sidebar "Start Your Engines."

Hyperprism
Minimum System Requirements
Mac: Power Mac 80; 16 MB RAM; Mac OS 7.6; Premiere-, VST-, or AudioSuite-compatible host.
PC: Pentium 133; 16 MB RAM; Windows 95/98/NT; DirectX host.

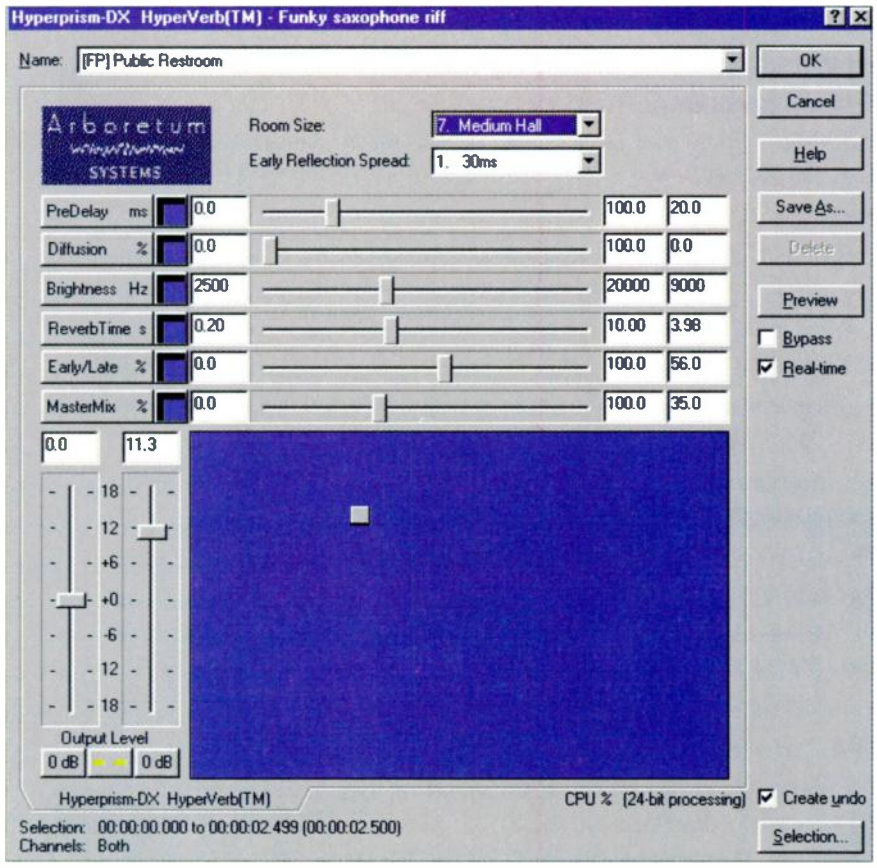

FIG. 1: *Hyperprism*'s blue window is an excellent way to control multiple parameters at once. Any parameter can be toggled on or off for real-time control.

AM I BLUE?

Hyperprism's most distinguishing feature is the large blue screen that serves as your control surface while working in the program (see **Fig. 1**). Unlike some other effects plug-ins, *Hyperprism* has no rows of knobs and buttons. Instead, *Hyperprism*'s work area is designed for manipulating control parameters by freely moving your mouse. The concept is simple: you determine which parameters you want to have under real-time control and how mouse movements in the window will affect their values, and then you perform the adjustments as a file plays back.

You can determine which axis of the blue window will initiate a change in value for each effect's parameters. For example, you could specify that vertical mouse movements alter the Feedback and Mix levels of a reverb and that horizontal movements change the Brightness level. For that same effect, you could set the Diffusion level to a fixed amount, unaltered by the mouse. The possibilities for linking parameters are endless—the Mix percent could increase as the Feedback decreases, for instance.

Equally important, you can change the range over which mouse movements alter an effect for each parameter individually. Using the example above, you might set the Feedback level to move between 20 and 30 percent. A sweep of the mouse from the top to the bottom of the screen would then only produce changes within that range. At the same time, the Mix values could be set to respond over a range of 0 to 100 percent for that same movement. (Additional blue window features, including parameter automation, are available in *Hyperprism* 2.1 for *HyperEngine*, *Hyperprism-TDM*, and *Hyperprism-VST* on the Mac.)

For manufacturer information, please see Contact Sheet, p. 163.

Of course, you don't have to adjust the effects in real time. You might want to use the blue window simply to set values that will remain fixed during playback. Just disable real-time updating and your mouse movements will have no effect. Nor do you have to use the blue window to alter values. All parameters have sliders that can be used for this purpose, and you can also simply type a specific number into the value window for any parameter. (Be careful not to hit Return after typing in a number; in the host programs I used, that immediately applied the effect to my file and returned me to the host's main window.) As in most effects plug-ins, there's a Bypass button that lets you compare the original file with a preview of the processed version.

Once you've tweaked an effect's parameters, you can save the configuration as a preset and reuse it on another file. In most host programs, you can preview the effect in real time only or apply it to the sound file to make destructive changes. PC programs such as Sonic Foundry's *Sound Forge* and Steinberg's *WaveLab* allow you to chain plug-ins (in series only), so you can easily build complex processes, limited only by your computing resources. Using BIAS's *Peak* 1.65 on the Mac, on the other hand, you can only run one effect at a time. In *HyperEngine*, however, you can use multiple plug-ins running in series.

▼

Especially interesting was a cross of a harp glissando with a woman's voice.

CAST OF CHARACTERS

The *Hyperprism* tool set is as large and comprehensive as any plug-in set I've seen. Though there are several similar effects—room and hall reverbs are separate effects, for example—most of the processes are unique. All of the effects offer a small (typically four or five) but generally useful number of presets that will get you started. I detected no problems with switching presets as a sound played back, which is a benchmark I often check when evaluating a plug-in.

The effects fall into several general categories. In the spatial group, you'll find AutoPan and Pan, which achieve the expected results; More Stereo, which alters the perceived width of an image and optionally applies a high-pass filter to your audio; and Stereo Dynamics, which has the ability to place a sound in a specific 3-dimensional location. I experienced a definite sense of expansion of the stereo field using the More Stereo effect, but I didn't find that the Stereo Dynamics "moved the input sound from front to back," as claimed by the Help file. (Of course, your own results will depend on many things, including your position relative to your monitors.) There's also an M-S Matrix effect, which can be used to tweak recordings made with mid-side miking techniques.

The time-based effects include Single Delay; Multi-Delay, which offers two independent delay lines with attenuation; and Echo. Echo adds a Feedback setting to the Mix and Delay

START YOUR ENGINES

HyperEngine is a free, stand-alone Mac application that ships with *Hyperprism* and can also be found in Arboretum's *Ray Gun* and *Ionizer* bundles. The program adds important new features to the *Hyperprism* package and provides a very efficient interface for its use. Though not a complete audio editor—there's not much more than basic cut and paste and some file navigation options, for example—it has many features that make working with the software more productive. In addition, some of the best-sounding *Hyperprism* effects are only available using this interface.

Hyperprism effects appear in *HyperEngine* with the same blue window that you find when using the plug-in versions, but as soon as you select an effect, you'll notice a new group of icons that appear at the top of the plug-in window (see **Fig. A**). These icons are used to draw parameter changes directly onto the screen. Circles, squares, rectangle, or doodles of any type can be used to automate effects parameters. Although you can't import pictures, as you can in U&I Software's *MetaSynth* (which Arboretum also distributes), you can get fancy and create some wild designs if you wish.

As a sound file plays, a small cursor races around the perimeter of your design to show you the current value assigned to a parameter. You can continue to draw after a sound starts to play, and you can even initiate playback by simply double-clicking in the drawing window. You can also save the drawings as part of a Reference document, which contains a sound file, the effects, and other associated data.

Every Reference document you open will have a "process list" displayed at the bottom of its window, below the waveform view. This list helps you manage your effects and has some very useful options, including enabling or disabling each effect individually. The list and the effects remain enabled even if you close the individual effect window, which helps keep the screen from getting too cluttered. You can drag and drop items in the list to change the order in which the effects are applied to the sound. It's also useful that you can apply the entire effects chain to a new file simply by using the Reassign function. And you can run the effects in real time or create a new file that includes all the processes you have loaded at any given moment. (Other versions preview in real time but process offline.)

FIG. A: *Hyperprism* for *HyperEngine* adds a number of new features, including parameter automation. All automation settings can be saved and reused with new source files.

One of the effects that are available only in *Hyperprism* for *HyperEngine* is Z-morph, which is similar to the Phase Vocoder except that it uses variable-pitched filters. You can also use two separate source files rather than only one stereo file. The Doppler effect is very effective—in no time I had sounds whizzing around my studio at high speed. Vari-speed uses an LFO to control the playback rate of your sound file. Try it with a very slow LFO speed on a drum loop or other repeating material; the variations in playback speed are extremely smooth.

Of all the versions I tried, my favorite effect is the HyperVerb, which is right up there with the best reverbs I've heard. There are numerous settings that you can use to find the perfect sound, as well as more than a dozen good-sounding presets to get you started. (All of the *Hyperprism* for *HyperEngine* effects include extensive presets.) I did run into some audio break-up when I tried to run the HyperVerb along with three other effects on my G3/266. But when I lowered the Quality settings by a notch, the problem went away. (You can adjust the processing and filtering quality settings to ensure that the program is fine-tuned for your system.) Of course, in non-real-time mode, this would not be a problem.

Hyperprism for *HyperEngine* offers a Play-Thru mode that allows you to send audio into the program for processing via any of the effects. This is a great feature because you can pass audio directly from one source, such as a CD player, "through" the effects and out to a DAT or other recording device. Unfortunately, my system is set for digital in and digital out using my Digidesign Audiomedia III card, and whenever I attempted to use this feature, the program reset my Control Panel to "Analog In." Arboretum has no fix for this problem as of yet, but is currently working on a cure.

Overall, *HyperEngine* offers the best environment for working with the *Hyperprism* plug-ins and is nearly an ideal interface for the experimentation and exploration that the program encourages. Though you may prefer the convenience of using plug-ins with your own favorite editor, you should definitely spend some time revving this engine.

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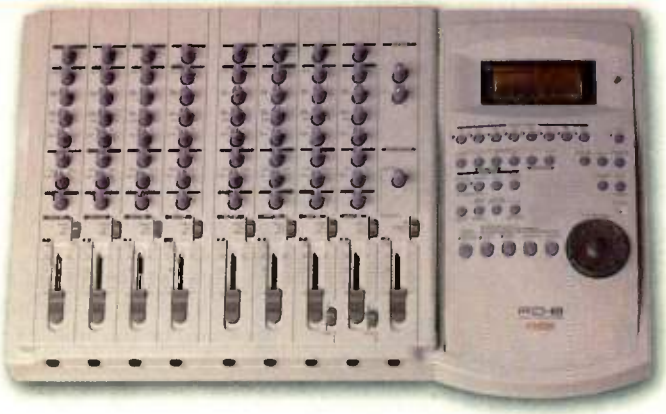

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HYPERPRISM VERSIONS

Hyperprism is available in plug-in versions for many host formats and can also be used with the dedicated *HyperEngine* audio application. Each version has a slightly different feature set. Here's a list of the current version numbers for the different formats and their list prices.

Hyperprism 2.1 for *HyperEngine* ($249)
Hyperprism-TDM 2.1 ($499)
Hyperprism Plug-in Pack 1.5.5 ($349), containing *Hyperprism-DX* 1.5.5 for Windows, Mac OS *Hyperprism-VST* 1.5.5, *Hyperprism-MMP* 1.5.5 for Premiere, and *Hyperprism-DAS* 1.5.5 for AudioSuite. *Hyperprism-DAS* can be used with Digidesign *Pro Tools* software and the Avid Media Composer 7.0 or Avid Xpress 2.0 nonlinear video editing systems.

Check with Arboretum for the minimum system requirements for each version.

Time options found in the delays. This group is especially efficient, using less than 10 percent of my Pentium II/266's resources. Among the dynamics processors is an excellent compressor with meters that provide a real-time display of input and attenuation levels. A limiter and noise gate round out the dynamics group.

Spectral alterations can be made with a large number of the effects. In addition to highpass, lowpass, bandpass, and band-reject options, the Harmonic Exciter can work wonders on your data. This effect, which sounds as good as the external hardware unit I use for similar purposes, adds new upper harmonics to the signal, creating more clarity and sparkle in your file. The Exciter was just the right effect for an old analog recording that needed some spiffing up. New to this version of the *Hyperprism* plug-in pack is an excellent bass maximizer, which is a popular tool these days. You can use it to really pump up the low end in a sound or just add a gentle boost to the bass frequencies. (And if there isn't much of a low end in your sound, the Maximizer can put one there!)

Several effects are intended to cross the spectral components of one sound with another. Both the Phase Vocoder

FIG. 2: The Ring Modulator effect multiplies a sound by a sine wave to add additional components to the signal.

and Z-morph (available in *HyperEngine* only) provide that capability. In the plug-in versions, the Phase Vocoder works a little differently than others I've seen. In this implementation, rather than specify two source files that you intend to cross, you vocode one channel of a stereo file with the other. (Other versions of the software allow you to use two mono or stereo files.) This is not as difficult as it might seem at first. For example, in *Sound Forge*, you can create a suitable stereo file by accessing the File Open menu, highlighting any two mono files, and enabling the Merge L/R to Stereo option. In *WaveLab*, you can simply copy one track of a stereo file and use it to overwrite one channel of another file. (Similar options are available in *Peak* and other Mac sound applications.) In either case, if you choose Sort by Size in your open file menu, you can see immediately which files are of similar lengths.

Once you have a suitable file, it's easy to adjust the various settings to get some interesting results. I played with different combinations for hours and ended up saving many of the sounds I created. Especially interesting was a cross of a harp glissando with a woman's voice; the result was a beautiful, choruslike effect that tracked the pitch of the harp.

Hyperprism includes both a Pitch Changer and Frequency Shifter but, sadly, no time-stretch effect. (There is one in the *HyperEngine* version.) You'll also find Tube Saturation and a Ring

(continued on p. 166)

ARBORETUM

Hyperprism multi-effects plug-in

$349

FEATURES ■■■■

EASE OF USE ■■■■■

AUDIO QUALITY ■■■■

VALUE ■■■■■

1 2 3 4 5

PROS: Large number of effects. Unique and efficient interface. High-quality sound.

CONS: Not all effects available in all versions. No printed manual.

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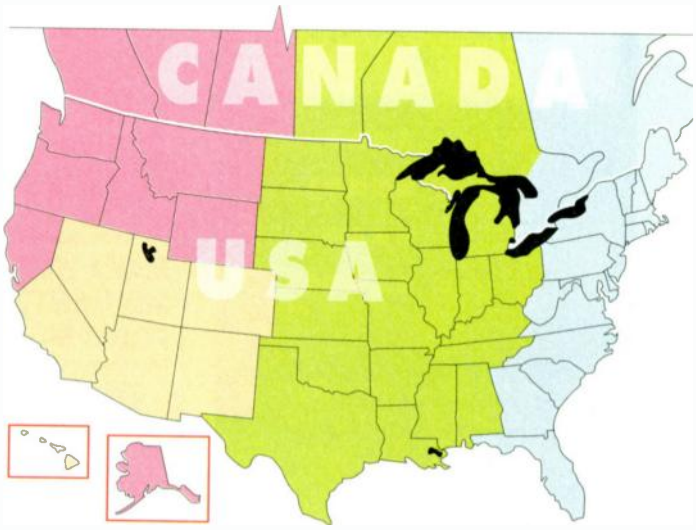

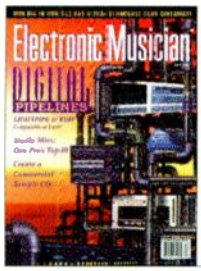

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(continued from p. 161)

Modulator (see **Fig. 2**). The Ring Modulator takes your sample and multiplies it by a sine wave, creating a host of new frequencies above and below that of the original sample. The oftentimes "metallic" result is a bit dated, in my opinion, but might be just what you need for a special effect or game sound. Among the remaining processes are chorus, flanger, phaser, tremolo, and vibrato.

MANUALS AND THINGS

Call me old-fashioned, but I really like having a printed manual on hand as I learn new software. Like many companies these days, Arboretum puts all the documentation on disk in HTML format. There is a "Contents" file among the several dozen individual files, and you can start your search for specific topics from there. But I'd rather have a real manual with an index, or at least the option of buying a copy from the company.

Overall, though, the manual topics and online help are mostly thorough and explain the parameters in every effect. Several of the new effects also have extensive application notes, but I would like to see far more examples of that type. There are also some inconsistencies between the names given to parameters in the documentation and the names that appear in the effects. For example, the Limiter page describes "threshold" and "level," but the help file and program use the terms "ceiling" and "volume."

Despite these minor complaints, *Hyperprism* is a powerful plug-in package that covers nearly all of the basic processing tools needed by modern musicians. It's especially useful if your audio editor doesn't have an extensive feature set or if you're looking for an additional high-quality toolkit to enhance your work. The unique blue window is an intuitive and effective way to modify parameters, and by allowing you to manipulate multiple values at once, it overcomes one of the biggest limitations of editing with a mouse. I also appreciate the painless copy protection used by the company: simply type in your serial number and away you go. Overall, *Hyperprism* would be a welcome addition to your audio projects.

Associate Editor **Dennis Miller** *lives in the snowy suburbs of Boston. Sometimes he envies the* **EM** *in-house California staff (but not often!).*

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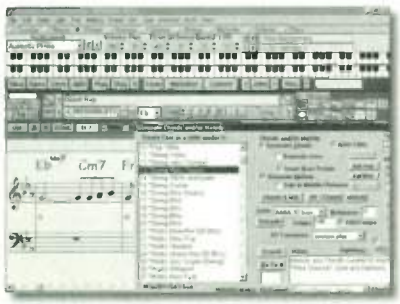

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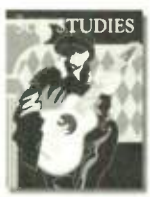

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MICHAEL BERRY

Pedalfects 1.1.1 (Mac)

By John Duesenberry

Michael Berry's *Pedalfects* ($100) turns any Power Mac into an elaborate real-time multi-effects stompbox with up to 8 simultaneous effects selected from 17 available types. Effects parameters can be set with the computer keyboard or mouse and controlled by LFOs or MIDI input.

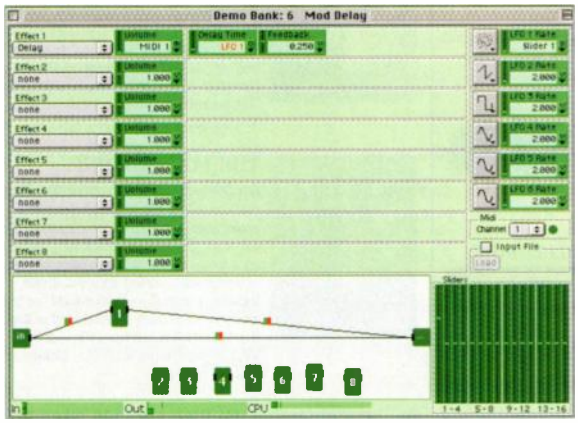

FIG. 1: The *Pedalfects* main window includes six sections for combining and controlling effects.

Pedalfects relies on Apple's Sound Manager for real-time audio input and output and requires OMS for MIDI control. A stereo AIFF file can also be used as the program's input source, and output can be recorded to a file in the same format. You can save *Pedalfects* presets in banks of 128 patches accessible through MIDI Program Change messages.

User Interface

The *Pedalfects* user interface consists of a single window divided into six sections (see **Fig. 1**). At the bottom left, a patching area called the Graph enables you to drag and connect eight numbered boxes representing the effects modules. Connections between modules can be mono or stereo, with pop-up volume and panning controls on the output of each. Above the Graph, a Control section lets you set the effect type and parameters for each module. **Figure 1** shows a simple patch in which the input signal is routed through a delay line, while the dry signal goes directly to the output.

To the right of the Control section, six LFOs offer sine, triangle, sawtooth, square, and random waveforms. A MIDI receive-channel selector and an input-file selector are located below the LFO section, and in the lower right, the Sliders section provides 16 sliders that can be assigned to any effect or LFO parameter. Finally, the Meter section along the bottom of the window displays input and output signal levels and CPU usage.

The parameters for each effect or LFO can be set directly by typing in numbers, or you can make adjustments with a miniature value slider to the left of the numerical field. You can also select a slider, a MIDI Continuous Controller, or an LFO as an alternate control source. In **Figure 1**, the delay line's volume, delay time, and feedback amount are controlled by MIDI controller 1, LFO 1, and a numerical setting, respectively. LFO 1's rate is controlled by slider 1 in the Slider section.

Pedalfects is laid out well, and it's easy to learn. The window, however, is sized to fit a very small screen. The onscreen sliders are tiny and difficult to manipulate, and the Graph area is also too small. It would be better if *Pedalfects* would let you resize its window and controls to take advantage of larger monitors. It would also be helpful if the values displayed were labeled with appropriate units such as Hz, milliseconds, or dB.

Performance

The effects available include reverb; filtering (highpass, lowpass, bandpass, parametric EQ, and allpass); normal and ping-pong delay; phasing and flanging; pitch shift and chorusing; dynamics processing (compression, gating, and threshold-triggered enveloping); and distortion. Taken as individual modules, most of these effects sounded reasonably good. The reverb, however, had a grainy, metallic quality, which I rather liked, but it didn't resemble a clean, professional-quality reverb.

The factory demo patches all involve five or fewer modules. To test the claim that *Pedalfects* can handle eight effects simultaneously, I concocted a patch involving four parametric EQ sections, a highpass filter, three reverbs, and all six LFOs. To make it work, I had to increase the memory partition to 24 MB. (The factory partition was 8 MB). This patch didn't quite swamp the CPU, but there was a noticeable degradation in responsiveness to mouse controls. This test was run on a 333 MHz G3 Power Mac, so it's a pretty good bet that, on slower machines, this patch would suffer audio degradation.

Wrap Up

You can download a demo version of the program (that does not save patches or record audio) from Michael Berry's Web site (www.nmol.com/users/mikeb/pedal.htm). The download package includes information on how to register. When the registration fee is received, a code is provided that enables all the features.

Pedalfects is a fun little program, but as I mentioned, there is room for improvement. The manual is sketchy at best, especially the effects reference section. I also encountered a number of bugs while reviewing the product, although most were quickly fixed by the author. Michael Berry seems quite dedicated in his support of the program, so I'm sure these problems can be overcome.

One good application for *Pedalfects* could be as an instructional tool in computer music or audio courses; the program's simple layout and flexible patching are ideal for novices. I doubt, however, that most audio professionals would be willing to tie up a Power Mac just for effects processing. I suggest that you download the demo version and see how *Pedalfects* performs on your machine.

Overall EM Rating (1 through 5): 2.5

CIRCLE #444 ON READER SERVICE CARD

STEINBERG

SPL De-Esser 1.0 (Mac/Win)

By Scott R. Garrigus

If you've ever recorded vocals, then you've surely encountered sibilance and the hassles that can accompany removing it. Traditional de-essing methods call for EQ and compression, but they can have negative side effects on the timbre of the original sound. Steinberg's *SPL De-Esser* ($199) plug-in claims to provide natural-sounding sibilance reduction with almost no negative side effects.

For manufacturer information, please see Contact Sheet, p. 163.

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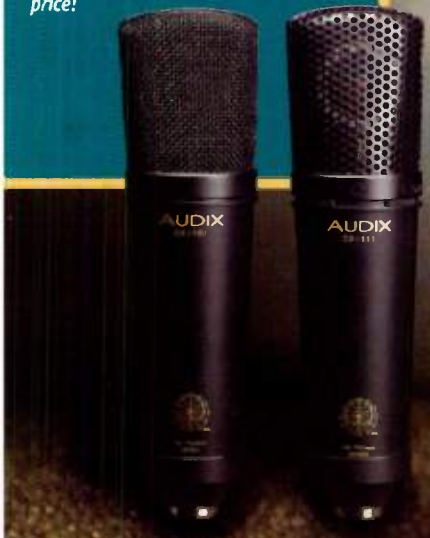


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QUICK PICKS

SPL De-Esser is modeled after its hardware counterpart of the same name and runs as a multiplatform plug-in for Steinberg's *Cubase VST* and *WaveLab* or for any DirectX-compatible application. The plug-in operates as a Channel, Insert, or Master effect in *Cubase VST* and as a Master Section effect in *WaveLab*. Operation via DirectX depends on the host application. Bypass, Mute, and Preset functions also depend on the host. All other functions are identical, no matter which program serves as host.

Steinberg's *SPL De-Esser* provides only a few controls; most of its functions are automatic. Even so, it does an excellent job of reducing excess sibilance.

Unusual Method

Instead of employing the usual EQ and compression methods, *SPL De-Esser* relies on phase inversion. The program scans the frequency spectrum of the original signal and identifies excessive sibilants or "s-frequencies." It then feeds them back into the original signal out of phase. The technique works well. The sound quality of *SPL De-Esser* is excellent, and even a high reduction setting has little or no effect on the sound of the original signal. I applied the process to a number of voice recordings (both spoken and sung), and I was unable to hear any side effects.

The program is easy to use. Just select the voice type (female or male) and set the amount of s-frequency reduction (0 to 10). The voice-type switches determine the center frequency of the bandwidth where *SPL De-Esser* starts its sibilant search (7 kHz for female; 6 kHz for male). The LED display shows how much each sibilant is reduced (0 to -20 dB) when you use the plug-in in real time.

Minimal Controls

An Auto Threshold function is also available. When active, *SPL De-Esser*'s input threshold automatically adjusts to the input signal level. When you disable Auto Threshold, the program uses a fixed threshold that the user's guide states has been "optimized for maximum level (normalized) signals." There's no way to adjust the actual threshold itself. You can, however, lower the level of the signal that you send to *SPL De-Esser,* which has the same effect as increasing the threshold value; in other words, less de-essing occurs. Conversely, if you increase the level of the signal, it has the same effect as lowering the threshold value: more de-essing occurs. The documentation offers a few tips for optimizing results.

Some recording engineers may dislike the lack of control that *SPL De-Esser* offers, but more control isn't necessary. The Auto Threshold function does a great job of following the input signal level, and with the right voice type selected, the program was able to reduce any sibilance that I threw at it. *SPL De-Esser* may not take a traditional approach to de-essing, but it definitely works, and works well.

Overall EM Rating (1 through 5): 4

CIRCLE #445 ON READER SERVICE CARD

ZENTECH

ChordMaster (Mac/Win)

By Mike Lawson

If you're a guitar player and you've ever experienced the frustration of trying to find a new chord fingering in a printed book, you'll love Zentech's *ChordMaster* CD-ROM ($29.99). The program runs on any 80486 or better PC and requires 8 MB free RAM for Windows 3.1 users or 16 MB for those running Windows 95 or 98. Macintosh

With more than 640 guitar fingerings and voicings, and a guitar sound library filled with AIFF files, Zentech *ChordMaster* is a path to musical enlightenment.

users will need to be running Mac OS 7.0 or higher and should have at least 4 MB of free RAM.

Enlightenment

With *ChordMaster,* you won't have to fight the binding of a guitar chord book so that it will lie flat. Instead, you can pop this handy disc into your computer's CD-ROM drive and instantly look up more than 640 guitar fingerings and voicings. You can also listen to each voicing strummed, arpeggiated, or as a combination of both. The program displays a grid featuring the 12 note names across the bottom and 20 variations of a given chord (such as flat, sus4, major, major 7th, and so on) along the side. Simply click on any of the positions in the grid, and you'll come up with the corresponding chord and its variation.

Pick a Chord, Any Chord

ChordMaster couldn't be easier to use. It offers clearly photographed examples of the fingerings on an acoustic guitar and stereo AIFF sound files that provide aural examples. In addition to the onscreen and audio examples, you can print each chord chart as an Adobe PDF file. (Adobe *Acrobat Reader,* for viewing and printing PDFs, is included with *ChordMaster.*) An extensive PDF help file is also included on the CD-ROM; the program is so easy to use, though, that I doubt you'll ever need to refer to it. The program gives you a library filled with guitar sound files for use with digital audio sequencers. The audio files were recorded as 16-bit, 22 kHz AIFF files and are organized in folders by key.

You'll get three fingering positions for 20 variations of 12 chord notes. You'll also see how the chord is written on the staff using standard notation. So if you're tired of playing the same chord positions and are ready to expand your fingering choices, pick up a copy of *ChordMaster.* It's already become a valuable part of my songwriting reference arsenal.

Overall EM Rating (1 through 5): 5

CIRCLE #446 ON READER SERVICE CARD

BOLDER SOUNDS

Bolder Pianos

By Jeff Obee

Show me a pianist who isn't particular about piano sounds, and I'll show you the door. *Bolder Pianos* ($129; CD-ROM) is a

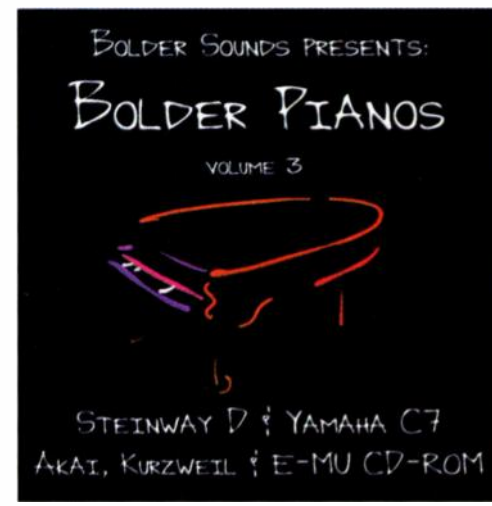
collection of grand pianos that should sate any pianist's ivory desires. This disc, the third volume in Bolder Sounds' sample-CD series, provides almost 400 MB of samples in Akai, E-mu, and Kurzweil K2000/K2500 formats.

Keying In

Stereo samples of Steinway D and Yamaha C7 pianos are proffered here, and each bank consumes between 25 and 32 MB of memory. A cap of 32 MB was set to allow room for as many perspectives as possible, and in order to accomplish that goal, some of the sounds are presented in ranges of less than 88 keys.

Each piano has a bank of two to eight variations: with or without effects, and bright or dark. There are also mezzo forte and fortissimo versions of each piano—each instrument takes on a very different character when sampled at fortissimo. There are looped and unlooped versions, but the looped files conserve only a small amount of memory and are about the same size as the unlooped files.

Rounding out the selection are the Velocity-switched pianos, which combine the mezzo forte and fortissimo sample sets; and lastly, a plucked piano, which Bolder Sounds sampled using a guitar pick, giving it a dulcimerlike quality.



The *Bolder Pianos* sample CD from Bolder Sounds gives you a well-rounded selection of classic grand-piano timbres.

Sounding Board

The Steinway D was recorded using two Neumann TLM 170 large-diaphragm microphones for maximum warmth. I liked the subtle, aged sound of this classic instrument. The mezzo forte version could work well in solo settings.

The Yamaha C7 was recorded using two AKG C 414s set in a figure-eight pattern (for room ambience) at about one meter apart, in a room with a solid wood floor. This piano is definitely brighter; the fortissimo samples have an especially sharp attack.

Both grands were run through Grace Design mic preamps. Bolder opted for Warm Room and Sweet Hall reverbs (a wet/dry mix of around 35 percent) for an unobtrusive ambience. Chorus was used on some sounds for a more overt, "electric grand" effect.

Testing 1, 2, 3...

I invited a world-class pianist to bring his Kurzweil PC88, listen to *Bolder Pianos,* and voice his opinions. He felt that all the sounds had an excellent, natural response. The fortissimo samples were too bright and "tack-like" for his tastes, although he thought that they would probably cut through well on live gigs. We concurred that the mezzo forte sounds of both grand pianos were warmer and more realistic.

The Velocity-switched pianos were a dubious success. We both felt that the change in the sound between mezzo forte and fortissimo was too obvious. Making use of the Kurzweil's three-tier Velocity keymaps would have made for more subtle gradations between samples.

The samples sound good on the upper end of the keyboard, where piano samples typically lose their realism. Under close scrutiny, I could discern some transition points between keymaps, and one keymap in the Yamaha samples (from C5 to D5) had a distinctly slower attack. I preferred the unlooped Steinway, due to its warmth and natural, resonant attributes.

Verdict

Overall, I enjoyed all these piano sounds and their inherent variations. Moreover, these samples sound good when summed to mono, which is handy for those who work with radio, TV, and multimedia.

Bolder Pianos is a fine collection at an excellent price—check it out for yourself with the MP3 files on the Bolder Sounds Web site. This disc may not end your search for the ultimate piano sound, but it can put some good sounds under your fingertips.

Overall EM Rating (1 through 5): 4

CIRCLE #447 ON READER SERVICE CARD

DON'T LET THE SIZE OF YOUR WALLET AFFECT THE QUALITY OF YOUR MONITORS

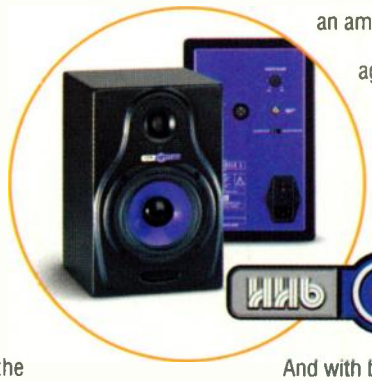
HHB CIRCLE 5

Combining great British pro-audio innovation with exacting manufacturing standards, the HHB Circle range of active and passive monitors delivers detailed, accurate and powerful sound at a new price point for precision studio monitoring.

The Circle 5 mid-field monitor features a revolutionary injection moulded cone, varied in thickness to minimise resonance, resulting in amazingly low distortion.

A controlled order crossover in both the active and passive models ensures that the Circle 5's sound is never tiring, even during long mixing sessions, and the complete absence of any limiting in the active bi-amp module means that the sound remains balanced and accurate at all listening levels.

Despite standing just over 10 inches high, the diminutive new Circle 3 packs an amazing punch and, again, is available in both active and passive versions.



HHB CIRCLE 3

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VIDEO and PRO AUDIO

DIGITAL MIXERS

MACKIE.

Digital 8 Bus Mixing Console

New

Well, it's finally here and just like the analog 8 bus a few years back, it's everything you've anticipated! Great sound quality, full recording and mixdown capabilities, motorized faders and an array of digital features geared to take you flying into the next century.

FEATURES-

- 48 channels of automated compression, gating, EQ and delay
- Built in 3-way meter display keeps you on top of your mix.
- Built-in meter bridge.
- Ultramix II automation for complete control, hook up an S-VGA monitor and you'll feel like you spent a lot more money.
- All functions can be automated, not just levels and mutes. Store EQ, reverb, compression, gating and even Aux send information.
- Fast SCENE automation allows you to change parameter snapshots on every beat.
- Reads Standard MIDI tempo maps, displaying clock info on the built-in position counter.
- Truly the cutting edge of mixing technology.

Panasonic

WR-DA7 Digital Mixing Console

New

Stop dreaming about your digital future, it's here! The Panasonic WR-DA7 digital mixer features 32-bit internal processing combined with 24-bit A/D and D/A converters as well as moving faders, instant recall, surround sound capabilities, and much more. Best of all, it's from Panasonic.

FEATURES-

- 32 Inputs/6 AUX send/returns
- 24-bit converters
- Large backlit LCD screen displays EQ, bus and aux assignments, and dynamic/delay settings.
- 4-band parametric EQ
- Choice of Gate/Compressor/Limiter or Expander on each channel
- 5.1 channel surround sound in three modes on the bus outputs
- Output MMC
- Optional MIDI joystick

TASCAM

TMD1000 Digital Mixing Console

New

You want to see what all the digital mixing buzz is about? The NEW TMD100 from Tascam will have you smilin' & automatin' in no time. It features fully automated EQ, levels, muting, panning and more in an attractive digital board with an analog 'feel'. Your digital future never looked, or sounded, so clear.

FEATURES-

- 4 XLR mic inputs, 8 1/4" balanced TRS inputs.
- 20-bit A/D D/A conversion, 64x oversampling on input, 128x on output.
- Store all settings, fully MIDI compatible.
- Optional **IF-TD1000** adds another 8 channels of TDIF and a 2-channel sample rate converter.
- Optional **FX-1000** Fx board adds another 4 dynamic processors and another pair of stereo effects.

MIC PREAMPS

Focusrite

Green 3 *"Voicebox MKII"*

New

The Voicebox MKII provides a signal path of exceptional clarity and smoothness for mic recording, combining an ultra-high quality mic amp, an all new Focusrite EQ section optimized for voice, and full Focusrite dynamics. The new MKII now includes a line input for recording and mixdown applications.

FEATURES-

- Same mic pre section as found on the Green Dual Mic Pre includes +48V phantom power, phase reverse, and a 75Hz high-pass filter. Mute control and a true-VU response LED bargraph are also provided
- EQ section includes a mid parametric band with frequency and gain control as well as a gentle bell shape to bring out the character of the voice.
- Dynamics section offers important voice processing functions of compression and de-essing combined with a noise reducing expander
- Single balanced Class A VCA delivers low distortion and a S/N ratio as low as -96dBu

EFFECTS PROCESSING

t.c. electronic
ULTIMATE SOUND MACHINES

Finalizer Plus

New

Improving on the multi-award winning Finalizer platform, The Finalizer Plus delivers an unprecedented level of clarity, warmth and punch to your mix. Inserted between the stereo output of your mixer or workstation and your master recording media, the Finalizer Plus dramatically rounds out your material, creating that "radio ready" sound.

FEATURES–

- Balanced Analog as well as Digital outputs including AES/EBU, S/PDIF, & TOS.
- 24-bit precision A/d & D/A Converters
- 5-band 24-bit stereo EQ
- Enhance - De-essing, stereo adjust or digital radiance
- Real-time gain maximizer
- Variable slope multi-band expander
- Multi-band compressor
- Word Clock Sync
- MIDI section useful for controlling sequencer fades or any of the Finalizer's parameters from a remote MIDI controller

Lexicon

PCM81 Multi-Effects Processor

New

The PCM-81 has everything that made the PCM80 the top choice among studio effects processors, and more. More effects, more algorithms, longer delay and full AES/EBU I/O.

FEATURES–

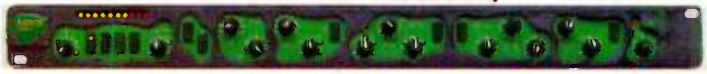
- 300 Presets include pitch, reverb, ambience, sophisticated modulators, 20 second stereo delays, and dynamic spatialization effects for 2-channel or surround sound applications
- 2 digital processors including Lexicon's Lexchip for the reverb and a second DSP engine for the other effects.
- 24-bit internal processing
- Dynamic patching matrix for maximum effects control.
- PCM card slot

EQUALIZERS

Focusrite

Green 2 *"Focus EQ"*

New



The Green 2 Focus EQ is suitable for a variety of applications combining a Focusrite equalizer section with a multi-source input section. Use it as a high-quality front end for recording applications or patch it into the send/return loop to upgrade a single channel of console eq, either way, it sounds great.

FEATURES-

- XLR & 1/4" inputs are similar to the Dual Mic Pre but have been adapted to cope with a wider range of levels.
- VU metering via a 10-LED bargraph
- EQ section derived from the Red and Blue range processors for superb audio quality.

COMPRESSORS

JOE MEEK

VC1 Studio Channel

got meek?

The Joe Meek Studio Channel offers three pieces of studio gear in one. It features an excellent transformer coupled mic preamp, a great compressor and an enhancer unit all in a 2U rackmount design. Find out why more and more studio owners can live without one.

FEATURES-

- 48V phantom power, Fully balanced operation
- Mic/Line input switch
- Mono photo-optical compressor
- High pass filter for large diaphragm mics
- Extra XLR input on front makes for easy patching
- Compression In/Out and VU/compression meter switches
- Twin balanced XLR outputs with one DI XLR output for stage use
- Enhancer In/Out switch and enhance indicator
- Internal power supply 115/230V AC

dbx

Blue Series 160S Stereo Compressor

The dbx 160S combines the best features of all the great dbx compressors in a well-built unit where the craftsmanship is as stunning as the engineering is innovative. This is truly a desirable compressor.

FEATURES-

- 127dB dynamic range • Program dependent "Auto", or fully variable attack and release
- Hard knee/OverEasy switchable.

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PHOTO - VIDEO - PRO AUDIO

MIDI HARDWARE

Mark of the Unicorn

MIDI Time Piece™ AV
8x8 Mac/PC MIDI Interface

The MTP AV takes the world renowned MTP II and adds synchronization that you really need like video genlock, ADAT sync, word clock sync, and even Digidesign superclock!

FEATURES–
- Same unit works on both Mac & PC platforms
- 8x8 MIDI merge matrix, 128 MIDI channels
- Fully programmable from the front panel
- 128 scene, battery-backed memory
- Fast 1x mode for high-speed MIDI data transfer

Digital Time Piece™
Digital Interface

Think of it as the digital synchronization hub for your recording studio. The Digital Timepiece provides stable, centralized sync for most analog, digital audio, and video equipment. Lock together ADATs, DA-88's, ProTools, word clock, S/PDIF, video, SMPTE, and MMC computers and devices flawlessly. It ships with "Clockworks" software which gives you access to its many advanced features and remote control of some equipment settings such as record arm.

OPCODE

Studio 64XTC
Mac/PC MIDI Interface

The Studio 64XTC takes the assorted, individual pieces of your studio-your computer, MIDI devices, digital and analog multitracks and even pro video decks, and puts them all in sync.

FEATURES–
- 4 In / 4 Out, 64 channel MIDI/SMPTE interface/patchbay with powerful multitrack & video sync features
- ADAT sync with MIDI machine control
- Simultaneous wordclock and Superclock output 44.1kHz or 48kHz for perfect sync with ADAT, DA-88 and ProTools
- Video and Blackburst in (NTSC and PAL)
- Cross-platform Mac and Windows compatibility

SAMPLING

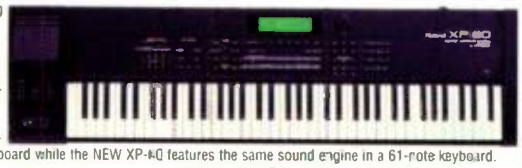
AKAI
S-Series
Rack Mount Samplers

Starting with 64X oversampling, Akai's S-Series Samplers use 28-bit internal processing to preserve every nuance of your sound and the outputs are 18- and 20-bit to ensure reproduction of your sounds entire dynamic range. These three new samplers add powerful capabilities, ease-of-use, expandability and affordability to set the standard for professional samplers.

KEYBOARDS & SOUND MODULES

Roland

XP60 & XP80 Music Workstations



The XP-80 delivers everything you've ever wanted in a music workstation. An unprecedented collection of carefully integrated features provide instant response, maximum realtime control and incredible user expandability. The XP-80 features a pro-quality 76-note weighted action keyboard while the NEW XP-60 features the same sound engine in a 61-note keyboard.

XP80 FEATURES–
- 64-voice polyphony and 16-part multitimbral capability
- 16 Mbytes of internal waveform memory; 80Mbytes when fully expanded (16-bit linear format)
- 16-track MRC-pro sequencer with direct from disk playback. Sequencer holds approx. 60,000 notes
- New sequencer functions like "non-stop" loop recording and refined Groove Quantize template
- Enhanced realtime performance capability with advanced Arpeggiator including MIDI sync and guitar strum mode and Realtime Phrase Sequence (RPS) for on-the-fly triggering of patterns
- 40 insert effects in addition to reverb and chorus
- 2 pairs of independent stereo outputs; click output jack with volume knob
- Large backlit LCD display

SR-JV80 Series Expansion Boards

New

Roland's SR-JV80-Series wave expansion boards provide JV and XP instrument owners a great-sounding, cost-effective way to customize their instruments. Each board holds approx. 8Mb of entirely new waveforms, ready to be played or programmed as you desire.

Boards Include–
Pop, Orchestral, Piano, Vintage Synths, World, Super Sound Set, Keys of the 60's & 70's, Session, Bass & Drums, Techno & Hip-Hop Collection.

KURZWEIL

K2500 Series Music Workstations

The K2500 series from Kurzweil utilizes the acclaimed V.A.S.T. technology for top-quality professional sound. Available in Rack mount, 76-key, and 88 weighted key keyboard configurations, these keyboards combine ROM based samples, on-board effects, V.A.S.T. synthesis technology and full sampling capabilities on some units.

FEATURES–
- True 48-voice polyphony
- Fluorescent 64 x 240 backlit display
- Up to 128MB sample memory
- Full MIDI controller capabilities
- 32-track sequencer
- Sampling option available
- Dual SCSI ports
- DMT Digital Multitrack interface option for data format and sample rate conversion (Interfaces with ADATs or DA-88s)

KORG

Trinity Series Music Workstations DRS

Korg's Trinity Series represents a breakthrough in sound synthesis and an incredible user interface. It's touch-screen display is like nothing else in the industry, allowing you to select and program patches with the touch of a finger. The 24MB of internal ROM are sampled using ACCESS which fully digitizes sound production from source to filter to effects. Korg's DSP based Multi Oscillator Synthesis System (MOSS) is capable of reproducing 5 different synthesis methods like Analog synthesis, Physical Modeling, and variable Phase Modulation (VPM).

FEATURES–
- 16 track, 84,00 note MIDI sequencer
- Flexible, assignable controllers
- **DRS (Digital Recording System)** features a hard disk recorder and various digital interfaces for networking a digital recording system configured with ADAT, DAT recorder and hard disk
- 256 programs, 256 combinations
- Reads KORG sample DATA library and AKAI sample library using optional 8MB Flash ROM board

TRINITY — *88 Weighted-key/Solo Synth*
TRINITY — *76-key/Solo Synth*
TRINITY — *61-key/Solo Synth*
TRINITY — *61-key*

**(Digital IF, SCSI, Hard Disk Recorder, and sample Playback/Flash ROM functions are supplied by optional upgrade boards)*

MONITORS

Hafler

TRM-8

Winner of Pro Audio Review's PAR Excellence Award in 1997, Hafler's TRM8s provide sonic clarity previously found only in much more expensive speakers. They feature built-in power, an active crossover, and Hafler's patented Trans•nova power amp circuitry.

- 45Hz - 21kHz, ±2dB
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MACKIE.

HR824

These new close-field monitors from Mackie have made a big stir. They sound great, they're affordable, they're internally bi-amped. "What's the catch?" Let us know if you find one.

FEATURES-
- 150W Bass amp, 100W Treble amp
- Full space, half space and quarter space placement compensation
- Frequency Response 39Hz to 22kHz, ±1.5dB

TANNOY

Reveal

New

The latest playback monitor from Tannoy, the Reveal has an extremely detailed, dynamic sound with a wide, flat frequency response.

FEATURES-
- 1" soft dome high frequency unit
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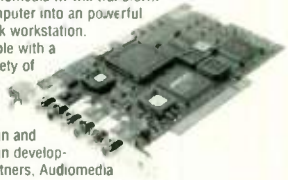
- Magnetic shielding for close use to video monitors
- Hard-wired, low-loss crossover
- Wide, flat frequency response
- Gold plated 5-way binding post connectors

SOUND CARDS

digidesign — AVID
A Division of Avid Technology, Inc.

Audiomedia III
Digital Audio Card

Working on both Mac and Windows OS systems, Audiomedia III will transform your computer into an powerful multitrack workstation. Compatible with a wide variety of software options from Digidesign and Digidesign development partners, Audiomedia III features 8 tracks of playback, up to 4 tracks of recording, 24-bit DSP processing, multiple sample rate support and easy integration with leading MIDI sequencer/DAW software programs.



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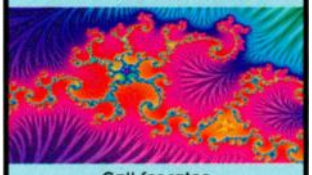

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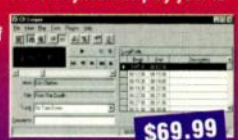

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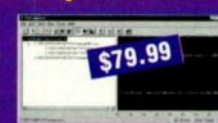

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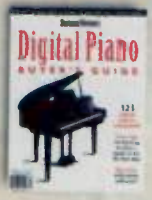

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FINAL MIX

By Larry the O

Everything All the Time

There's simply no arguing the truth of the axiom "Less is more." Having fewer elements in a mix allows for greater openness and clarity. Even the great producer George Martin states that he questions the necessity of every note when he scores. (See "Production Values: To Sir with Love" in the February 1999 issue of **EM**.)

But is that point of view the only truth in the matter? I've also heard it said that nothing succeeds like excess; or, as Bob Weir sang, "Too much of everything is just enough." What about Phil Spector's famed "Wall of Sound"? Or Steely Dan's microscopic approach to record making? The late Frank Zappa once told me in an interview that he'd used 3,000 inches of splicing tape (at one inch per splice) for his London Symphony Orchestra album—and that was all razor-blade editing of the digital multitrack master!

Want more examples? Multiple Oscar-winning sound designer Gary Rydstrom recorded and edited gun, bullet, ricochet, and explosion sounds individually for the battle ambiences in *Saving Private Ryan,* and the end product had amazing realism and impact, as well as incredible sonic density. Some facilities have installed film-mixing consoles with more than 300 inputs to accommodate just such a working style. And, lest we forget, the Beatles, under the discriminating production guidance of Martin, often piled sound upon sound, including the four passes of orchestra overdubs on "A Day in the Life."

Perhaps this latter example is the one that best makes my point that there is a time and place for a simple, spare approach; there is also a time and place to find out what happens when you turn the knob up to 11. For **EM** readers, this statement has never been more true than now, when many of us are able to afford excellent sounding tools that can largely remove the ticking of the studio clock, the sword of Damocles that imposes time pressure on the creative process.

If you have your own digital multitrack system (whether tape or disk based), a decent microphone and mic preamp, an effects box, and some good monitors, why shouldn't you record take after take until you get something the way you want it to be? Why shouldn't you pile on track after track to see what it sounds like? You don't have to *use* all that stuff just because you recorded it. Heck, you don't have to use *any* of it.

You have the ability to lock up more than a dozen ADATs to run in sync together, so why not do it? Three tracks for vocal takes? Why not 30? Don't worry if you end up with a lot of tracks to mix; you can always cascade digital mixers to get more channels.

And what about MIDI? Today's interfaces can give you more than a hundred channels, and those channels are there to be used. Consider the drum bank of a typical synthesizer: each drum or percussion instrument is mapped to a single key, and the only per-sound volume control is Velocity, which sometimes is not enough. In some very fussy situations, I've given a MIDI channel over to each drum or percussion instrument to give me the most comprehensive control.

Plainly, there are times when more is more. Most people, when mixing, will remove an element that doesn't sound as though it works with the piece. Although this does mitigate the undesirable element's negative impact on the overall zeitgeist, it can leave a lot of holes. By simply adding another element to cover the bogie, the bad element is no longer heard, yet the combination of elements yields a greater depth to the texture of the mix. The process is just like cooking: if the soup tastes bad, you keep adding to it until it tastes good. And if it never tastes any better, just throw it away.

Larry the O *is no longer a San Francisco-based musician, producer, and engineer. He is still a sound designer at LucasArts Entertainment. He's been grooving heavily on microphones lately.*

ILLUSTRATION BY STEVE CURL

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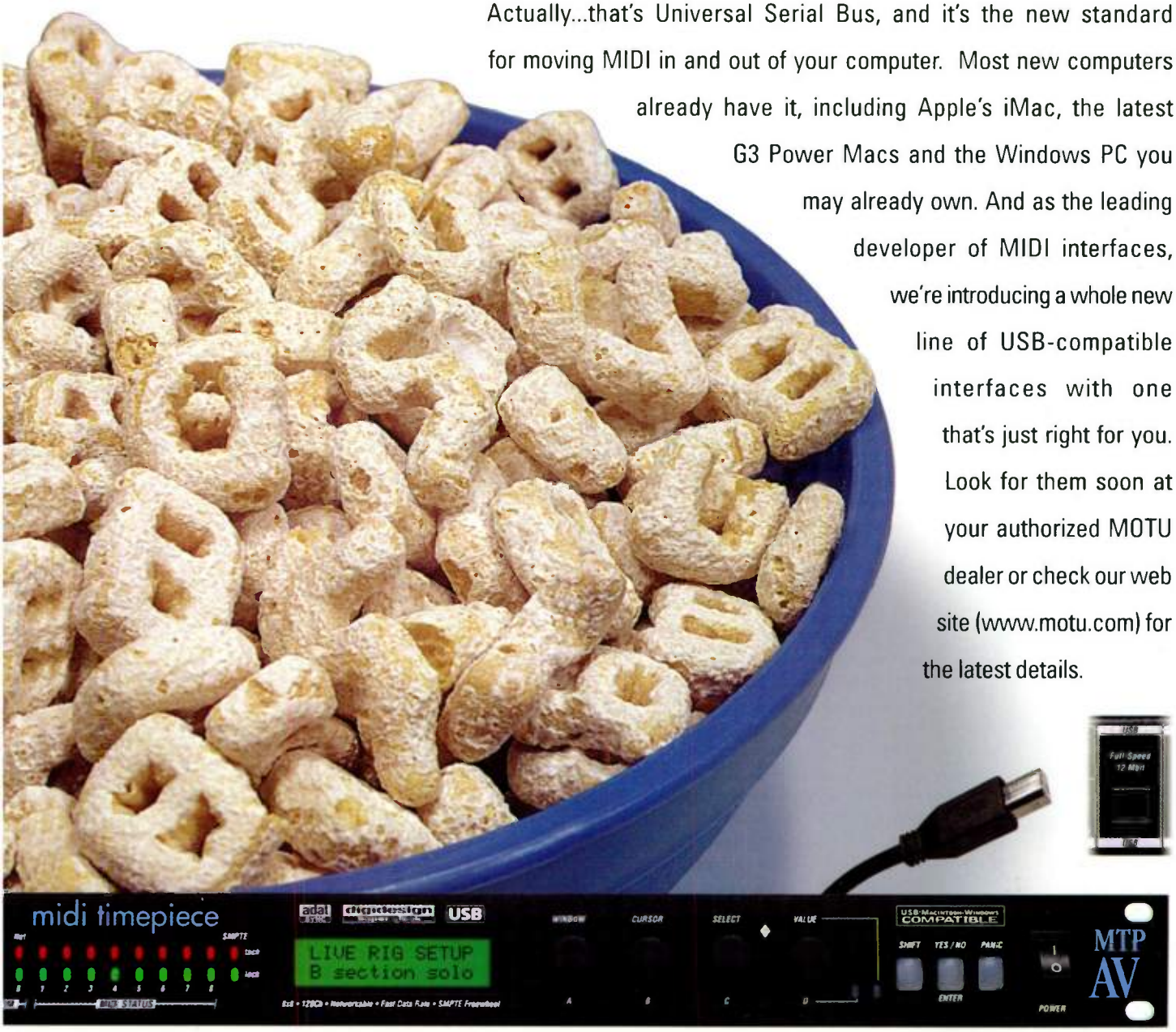

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