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September 1999

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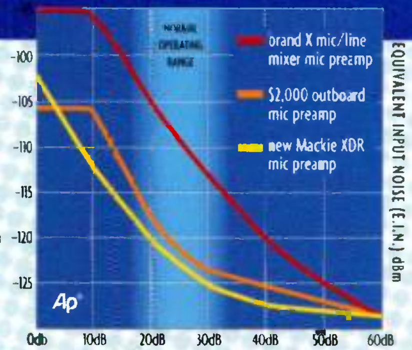
Two years in the making, XDR™ Extended Dynamic Range the pristine sonics and awesome specs of high-end ou

If your hype alarm is going off, we can't really blame you.

The proof is in the listening. Visit your nearest Mackie dealer and audition the XDR™ preamp design in our new VLZ PRO Series compact mixers. Use the most expensive microphone they have. Compare XDR to ultra-expensive outboard mic preamps. Compare it to our compact mixer competition. Bring your golden-eared audiophile friends. We think you'll be amazed. We honestly believe that you've never before heard a mic preamp this good.



Ⓐ A new standard for low noise in the critical +20 to +30dB operating range.

If money is no object, don't read any farther.

If you can afford $1000 a channel for outboard "audiophile" mic preamplifiers, DO IT! Because no matter how much you spend on a microphone, its ultimate performance depends on how it interacts with the preamp it's plugged into.

Yes. We openly admit it: Many high-end mic preamps can effortlessly amplify the slightest sonic nuance, creating an aural panorama that's breathtakingly realistic, excitingly vivid and truly 3-dimensional in scope. For years, they've provided fidelity that just hasn't been possible with the "stock" mic preamps built into mixing consoles.

Until now.

A massive R&D initiative.

We can confidently say that no other company in the world has spent the sheer number of engineering hours —and $250,000 in R&D costs that we just did on a single new microphone preamplifier design.

The XDR team started with blank paper, concerned only with matching or exceeding the performance of $500 to $2000-per-channel esoteric preamps. They went through hundreds of iterations and revs and spent countless hours subjectively listening (and arguing). They started all over again several times. They scoured the world for rare parts. Then they spent more time critically listening and evaluating the

NEW!

1604-VLZ PRO

16x4x2 • 16 XDR™ preamps

60mm faders • 16 mono chs.. • 4 sub buses • main L/R • 3-band EQ with sweepable midrange (12kHz & 80Hz shelving, 100Hz-8kHz mid) • 18dB/oct.@ 75Hz low cut • 6 aux sends per ch. • Constant Loudness pan controls • 4 stereo aux returns • RCA tape inputs & outputs • 16 channel inserts, 16 high-headroom line inputs • 8 direct outs • TRS balanced outputs • Switchable AFL/PFL Solo • Ctl Room/Phones matrix with Assign to Main Mix & separate outputs • Ctl Rm/Phone level control • 12-LED metering plus Level Set LED & RUDE Solo light • Aux 1 & 2 Pre/Post • Aux Send master section w/level controls • Solo buttons with LEDs • Stereo Aux Return assign section with EFX to Monitor & Main/Submix assign • built-in power supply • solid steel main chassis • BNC lamp socket • Rotatable I/O pod allows 5 different physical configurations

NEW!

1402-VLZ PRO

14x2x1 • 6 XDR™ preamps

60mm faders • 6 mono & 4 stereo chs. • 3-band EQ @ 12kHz, 2.5kHz & 80Hz • 18dB/oct.@ 75Hz low cut • 2 aux sends per ch. • Constant Loudness pan controls • 2 stereo aux returns • RCA tape inputs & outputs • 6 channel inserts, 6high-headroom line inputs • XLR & TRS balanced outputs • switchable +4/mic level output • ALT 3-4 stereo bus • Switchable AFL/PFL Solo • Ctl Room/Phone matrix with Assign to Main Mix & separate outputs • Ctl Rm/Phone level control • 12-LED metering plus Level Set LED & RUDE Solo light • Aux 1 Pre/Post • EFX to Monitor • sealed rotary controls • built-in power supply • solid steel chassis

NEW!

1202-VLZ PRO

12x2x1 • 4 XDR™ preamps

4 mono & 4 stereo chs. • 3-band EQ @ 12kHz, 2.5kHz & 80Hz • 18dB/oct.@ 75Hz low cut filter • 2 aux sends per ch. • Constant Loudness pan controls • 2 stereo aux returns • RCA tape inputs & outputs • 4 channel inserts, 4 high-headroom line inputs • XLR & TRS balanced outputs • switchable +4/mic level output • ALT 3-4 stereo bus • Ctl Room/Phones matrix with Assign to Main Mix & separate outputs • Ctl Rm/Phone level control • 12-LED metering plus RUDE Solo light • Aux 1 Pre/Post • EFX to Monitor • sealed rotary controls • built-in power supply • solid steel chassis

WRH

IVE ESOTERIC MIC PREAMP SOUND

mic preamp circuitry gives our new VLZ PRO™ Series board microphone preamplifiers.

design with every high-end microphone you can think of. Then they brought in veteran recording and live sound engineers for more exhaustive listening tests.

What we ultimately ended up with is not just an awesome sounding design. XDR is also **a)** highly resistant to damage caused by "hot patching" (caused by routing a phantom powered mic through a patch bay) ; **b)** remarkably independent of cable-induced impedance variations; and **c)** able to reject extremely high RF levels without compromising high frequency response.

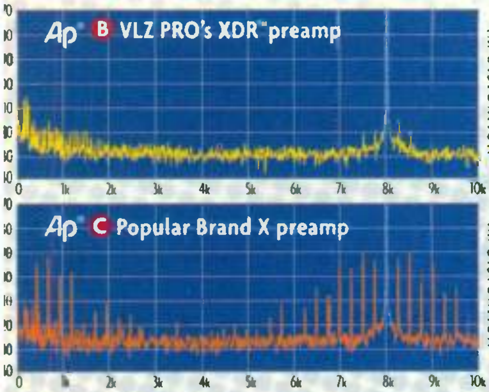
WARM,DETAILED SOUND
0.0007% THD
NEAR DC-TO-LIGHT BANDWIDTH

XDR

OVER 130dB DYNAMIC RANGE FOR 24-BIT, 196kHz SAMPLING RATE INPUTS
ULTRA-LOW IM DISTORTION & E.I.N. AT NORMAL OPERATING LEVELS
IMPEDANCE INDEPENDENT
BEST RF REJECTION OF ANY MIXER AVAILABLE

Rejecting RFI without tuning out sound quality.

Because a mic preamp must amplify faint one millivolt input signals up to a thousandfold (60dB), its rectification components can also pick up radio frequency interference (RFI) from AM and FM stations, cell phones and pager transmitters —even microwave ovens—and amplify them to audible levels.



XDR vs. Brand X FFT analysis of mic preamp Intermodulation Distortion. Mixer trims at 30dB, 0dB at inserts. The white spike at 8kHz is the fundamental tone used to "generate" the surrounding distortion artifacts—which the Brand X mic preamp has far more of.

We assaulted RFI on three fronts. First, we incorporate bifilar wound DC pulse transformers with high permeability cores that reject RFI but don't compromise audible high frequency response. Second, we carefully-matched high-precision components for critical areas of the XDR preamplifier. Third, we direct-coupled the circuit from input to output and used pole-zero-cancellation constant current biasing (which also avoids increased intermodulation distortion at high signal levels).

Bottom line for the non-technical: Our new VLZ PRO Series has the best RFI rejection of any compact mixers in the world. Period.

Controlled Interface Input Impedance.

If a mic preamp isn't designed right, it will actually sound different depending on the impedance of the microphone and the cable load!

XDR's Controlled Interface Input Impedance system accepts an enormous range of impedances without compromising frequency response. Whether the mic/cable load is 50 ohms, 150 ohms or 600 ohms, XDR mic preamp frequency response is down less than one tenth of a dB at 20Hz and 20kHz!

Ultra-low noise at "Real World" gain settings.

Many mixers that tout low E.I.N. (Equivalent Input Noise) specs can't deliver that performance at normal +20 to +30dB gain settings. Graph A on the other page charts E.I.N. versus gain level for our new VLZ PRO Series vs. a major competitor's mic/line mixer preamps and a "status" outboard mic preamp retailing for about $2,000. As you can see, our XDR design maintains lower noise levels in the critical +20 to +30 gain range than either competitor.

There's still more:

- **0.0007% Total Harmonic Distortion.** The lowest ever in any compact mixer.
- **Flat response.** Not only are XDR mic preamps flat within a tenth of a dB across the bandwidth of any known microphone, but are also only 3dB down at 10Hz and 192kHz!
- **116dB CMRR** 20Hz to 200kHz and above.
- **Super-low intermodulation distortion** at very high operating levels (charts B&C at left) thanks to instrumentation-style balanced differential architecture, linear biasing and use of DC-coupled pole-zero-cancellation constant current that frees the mic preamp from power supply fluxuations.

We could go on and on this way. But like we said at the start of this ad...

Hearing is believing.

Visit your nearest Mackie Dealer. Select a really high-quality condenser mic and try out the new 1604-VLZ PRO, 1402-VLZ PRO and 1202-VLZ PRO Think of them as expensive esoteric mic preamps... with really excellent compact mixers attached.

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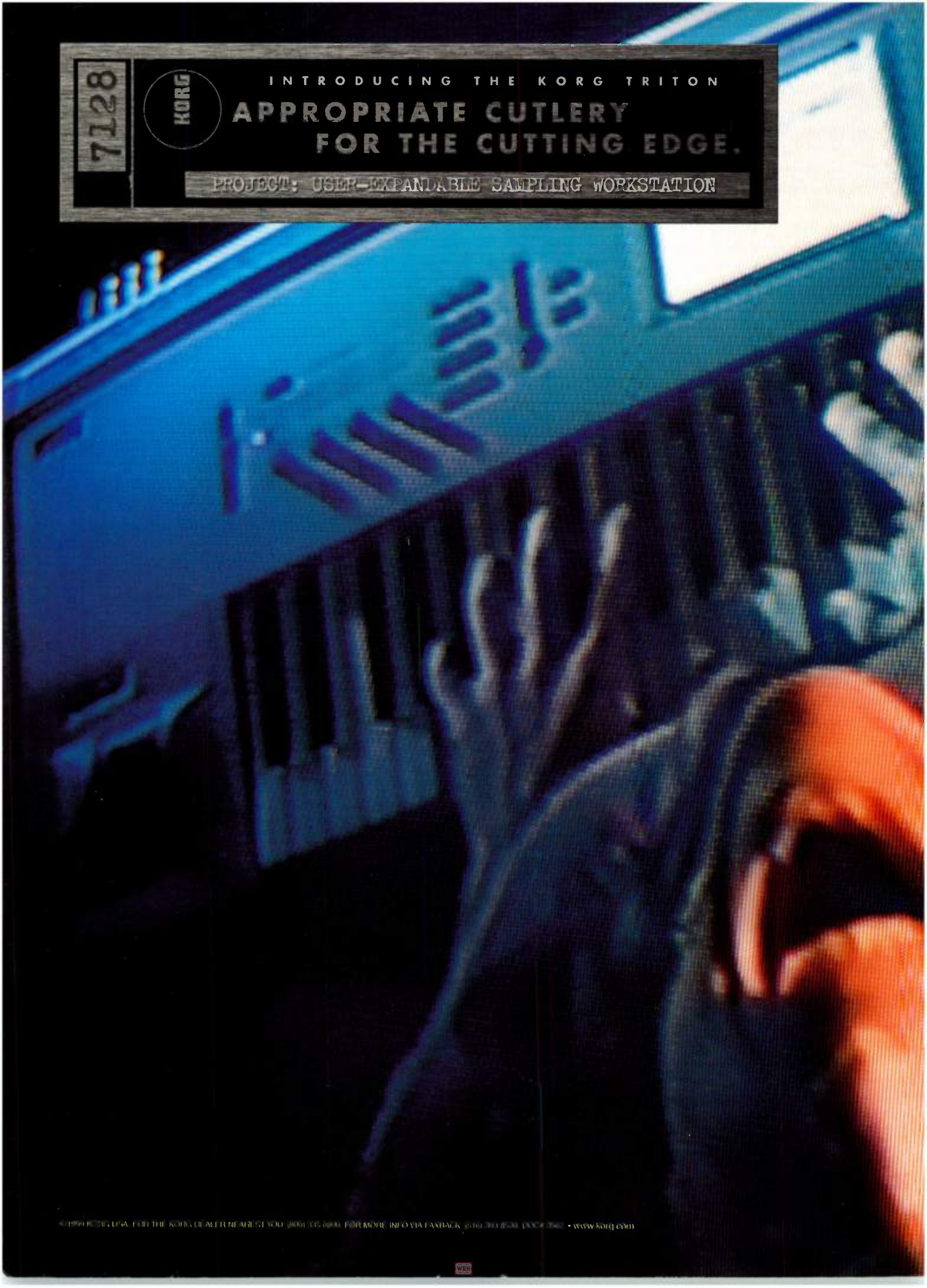
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All you need is here. Beneath that glowing TouchView screen beats the heart of a technical behemoth. Just put your hands and eyes on your instrument. And your creative soul will follow.
Recapturing the creative fire, as it turns out, isn't nearly as difficult as you thought. You need only to give it a Tri.
KORG
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I N S I

FEATURES

DEPARTMENTS

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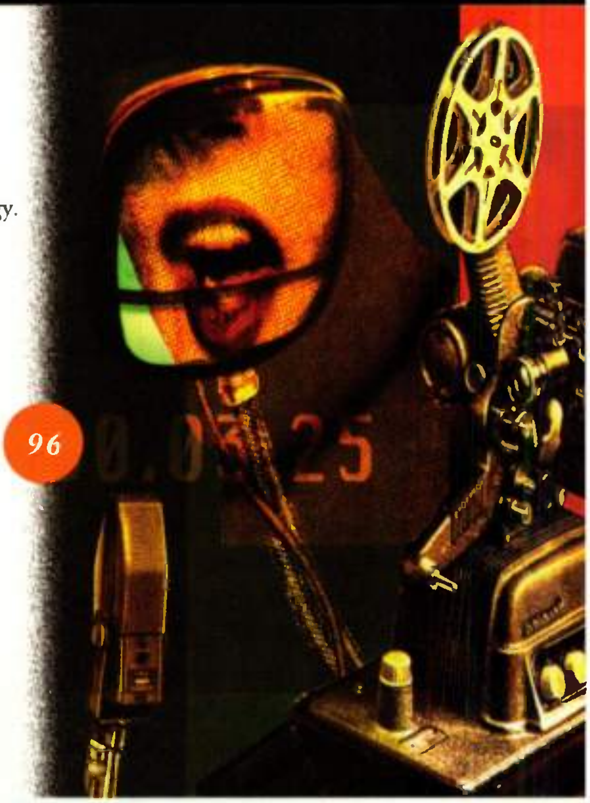
COLUMNS

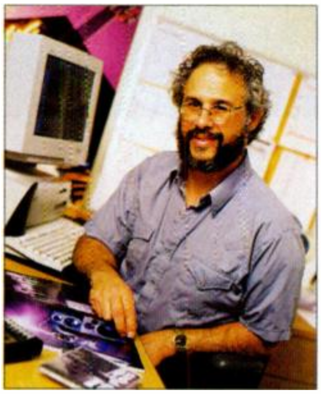


REVIEWS

Cover: Photography by Philip Kaake

What's in a Name?

It's easy to distinguish between shareware and commercial demo software.



ANTHONY PIDGEON

If there is a heaven, surely it holds a place for the authors of freeware and shareware. It's encouraging that in our often self-centered and jaded society, some programmers trust users to pay tomorrow for software that they can use today—and others even give away their wares. I promptly pay the fee for any shareware I use. Conversely, if I check out a program and decide not to use it, I don't pay for it. That's the beauty of shareware.

However, according to one well-informed developer, the Association of Shareware Professionals changed its definition of shareware several years ago to include programs that, for instance, have certain key features (including file saving) disabled. I think that this misses the point of shareware, and that the new definition unnecessarily blurs the line between shareware and commercial demos.

What's in a name? For **EM**'s purposes, a program must meet all of the following criteria to qualify as shareware:

1. Payment is expected, but enforcement is strictly on the honor system.

2. The program is usable as is, with all the essential features that are required to make practical use of the program fully operational. You might get a code that unlocks additional features when you pay your shareware fee, but if fundamental features (such as the ability to save a file) are crippled in the unregistered program, or certain features work only part of the time, it is not shareware; it is a demo.

3. You can use shareware indefinitely without paying, though you might get reminder messages every time you launch the software. If software "times out" (that is, expires after some period of time) and is thereafter unusable until you pay for it, it is not shareware; it is a demo.

4. Shareware is not physically copy protected. However, if you copy a registered version, you may have to reenter a code to authorize the copy.

Shareware authors may, of course, offer incentives to register, such as supplying free documentation, giving you access to additional features, providing tech support, and including you on a mailing list. Payment for use of shareware might be in the form of money, a postcard, a copy of your music, or even a credit. But there is no penalty if you don't register.

As many denizens of the Web can attest, a wealth of shareware programs that meet all of our criteria is available. At sites such as Shareware Music Machine (www.hitsquad.com/smm), you can find lots of "traditional" music shareware, freeware, and product demos for a variety of operating systems. Did you know, for example, that music shareware exists for OS/2? I didn't before visiting this site.

Mind you, there's nothing wrong with commercial software demos. They allow you to check out at least some features before you buy, which is excellent for all concerned. However, labeling demos as shareware is just plain sophism. When we discuss shareware in **EM**, we will include only the real thing.

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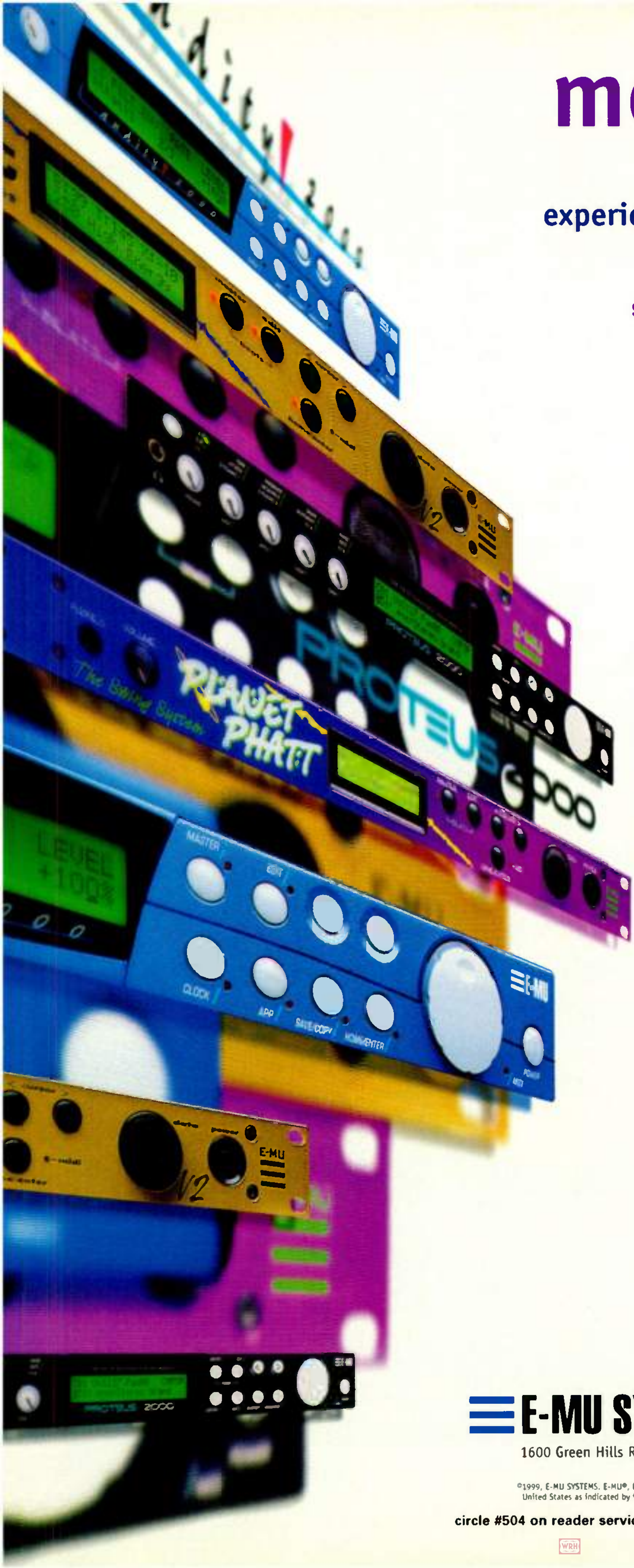

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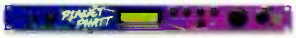

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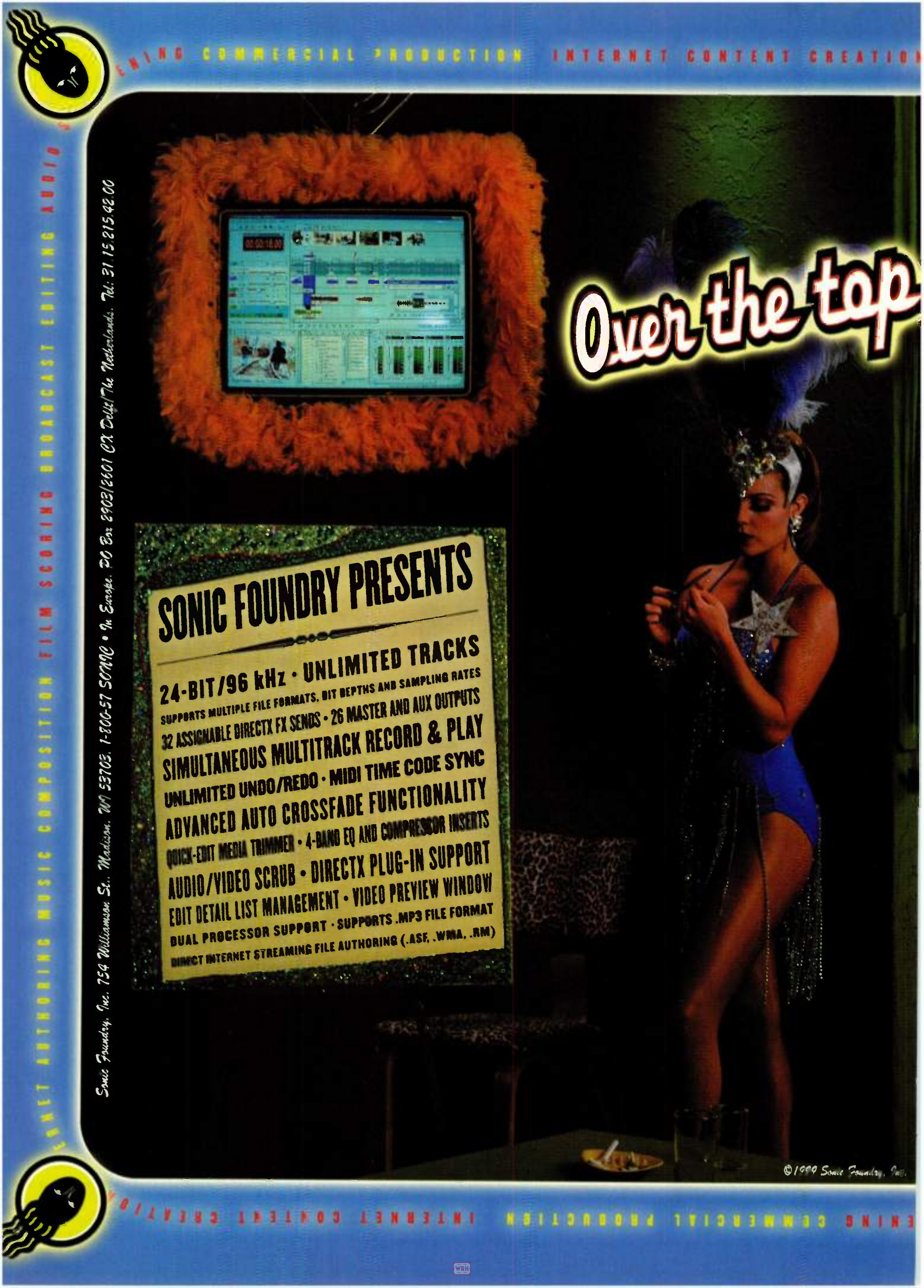

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PAUL CORIO

SETTLE THE SCORE

Scott Garrigus's article "Scoring on the Web" ("Desktop Musician," July 1999) is interesting, and I would like to bring up a few points pertaining to music notation software.

More than ten years ago, a committee of academics, programmers, and music companies was formed to establish standards for music notation software, so that software from any manufacturer could import files from other manufacturers' programs. The proposed file format was called Notation Information File Format (NIFF). This is similar to MIDI in the music world, or to ASCII in the word-processing world.

It is important to note that the NIFF format has gone absolutely *nowhere* since its proposal. Why? I do not know. It could be that the largest companies, such as Coda Software *(Finale)*, fear that if there were a working standard, they would sell less software.

Surely, with the success of MIDI, there is little doubt that having a standard interface is beneficial to users and manufacturers alike—financially, creatively, and otherwise. This is even more important in the comparatively small specialty area of music notation. The reality is that there are precious few dollars going into music notation development; therefore, having a standard interface would benefit everyone.

I am a professional music copyist and engraver, and have been since before the days of computing. Right now, in the present state of the art, it is still usually faster to re-input music from scratch than to try to import music from another program. Can you imagine this being acceptable in the text-typesetting world? Having the NIFF interface would make efficient file translation possible.

I feel that publishing music notation on the Web is not really that difficult—after all, it can be done now with any program that can save to Adobe PDF (Portable Document Format) files. Not only that, any software that is commercially available *could* provide a limited "reader" format as a plug-in for the Web; certainly this is done by many other manufacturers.

The typical professional musician, composer, or songwriter makes a very small profit on sales of printed music. Manufacturers should be making distribution of printed music easier and cheaper, and they could do this by adopting a robust, exchangeable, open-ended approach to music notation.

Ernie Mansfield
ernieman@dnai.com

Author Scott Garrigus replies: *To my knowledge, NIFF is still alive but not really kicking. A small number of software applications support the format, including* MidiScan *and* PianoScan *by Musitek, MusicWare's* Nightingale, *and Noteheads'* Igor. *There may be more, but those are the ones I know of. I'm really not sure why the more prominent notation software developers are not supporting the format. I agree that having a standard notation file format supported by all the major applications would certainly benefit all users—and the manufacturers, too. Just look at how widespread the Standard MIDI File format has become. And, like SMF, NIFF can be used by anyone without any set payment or royalty fees. As a matter of fact, there's a free, public-domain, platform-independent NIFF SDK (software developer's kit) available for anyone who*

MEET THE "ULTRA-TECHNOLUST GEAR GIVEAWAY" WINNER!


SARAH ESSEX

Congratulations to Arthur Winer, winner of the EM Ultra-Technolust Desktop Studio Giveaway! Winer's name was drawn on June 15, 1999, as the winning entry for a prize totaling $22,864.55 in desktop studio gear. Winer is a 31-year-old student and musician living in Brooklyn, New York. After working as a book editor and business writer for many years, Winer decided last year to change directions entirely and return to school for music. He is currently finishing his Master's degree in Music Technology at New York University and hopes to find work as a recording and mastering engineer.

Winer is also the founder and songwriter of the New York–based alternative rock band MacArthur. The band's eponymous debut CD was released this summer on Winer's new independent label, Canaveral Skies Music. For more information about MacArthur or Canaveral Skies Music, contact Winer by e-mail at awiner@earthlink.net.

The EM staff congratulates Arthur Winer on winning the sweepstakes and thanks all the manufacturers that participated: Adaptec, AKG, APC, Apple, Arboretum, ART, Event Electronics, Iomega, Korg USA, Mark of the Unicorn, Omnirax, Panasonic, QuikLok, Seagate, and Yamaha.

wants to implement NIFF in their applications. There's a link to it on the following Web page: http://mistral.ere.umontreal.ca/~belkina/NIFF.doc.html. You may also want to get in touch with Professor Alan Belkin, a special advisor and part of the team that created NIFF. You can find Professor Belkin's e-mail address on that Web page as well.

As far as using Adobe's PDF files to exchange music notation over the Internet, it's certainly an excellent solution for one-on-one musical collaboration. But the format doesn't allow you to actually edit musical notation, which puts a bit of a damper on exchanging files that way. It's not a secure format, either, so selling sheet music as PDFs can be cumbersome.

If NIFF were to become as widely accepted as the SMF format, music notation would be just as easy to deal with as MIDI data. That would certainly make our musical lives a lot more productive.

STRESSED CHIPS

Maybe you guys can help me out. I have been reading about the wave of new converters that are popping up all over the place, especially the ones that support sample rates of 96 kHz, 192 kHz, and up. I found out that they have the same chip as 44.1 and 48 kHz converters, and that they (96, 192, etc.) are just overclocked. Is this true? Is this possible? As an **EM** reader, I would like to know more about it. For example, is the chip "physically" altered, or is this a change on some kind of embedded "operating system" inside or around the chip?

Lincoln Martins
linclink@gate.net

Lincoln—Thanks for your letter. I am really interested in this question, so I asked Kurt Hebel, vice president of engineering at Symbolic Sound, for his explanation. According to Hebel: "The DSP chips used in A/D/A converters are not 'overclocked' as a rule. In order to improve the performance of a chip and allow it to operate at a higher sampling rate or greater bit depth, the manufacturer must make improvements to the chip's design. For example, you can speed a chip up (to a point) by making it smaller. This process is called 'die shrinking'—perhaps you've heard reports of Intel shrinking its chips prior to the release of a new generation of Pentiums. A manufacturer can also find clever new ways to improve the logic or architecture of the chip. Modern DSP chips have their own memory onboard, and the logic that runs the chips is always improving. You can also speed up a chip by using better transistors—though not necessarily more of them.

"Overclocking, which is not a safe process, is typically employed by computer users who wish to get higher speeds from their CPUs than those recommended by the manufacturer. Also, overclocking should not be confused with 'oversampling,' which is a technique employed in nearly every modern A/D/A converter."—Dennis M.

BURNING QUESTION

I have been making CD-Rs in my home studio and seem to be constantly trying to find a way to make my CDs louder. I have noticed that CDs I have had duplicated from DAT elsewhere are significantly louder than CD-Rs made from the same tracks in my studio. I did not pay for additional mastering, yet these CDs are at least 4 dB louder.

I am wondering if my commercially duplicated CDs are louder because of additional processing done by the CD manufacturer, or whether I need better CD-mastering software. If commercial CDs are louder due to additional processing gear I do not have, can you give me any suggestions on how I can squeeze a few more decibels out of my CD-Rs?

Chaz Ipson
gorbyrun@stargate.net

Chaz—It's probably worthwhile to ask the mastering engineer at your replicator if he or she modified your master before replication. In my experience, replicators usually contact the client before doing anything to a master; but I've also heard a number of stories to the contrary. Sometimes replicators are handed a do-it-yourself CD-R master that has problems, and they have the option of fixing the problems (if they're minor) or sending the original back to the client to be redone. Problems can include high block-error rates (BLER), clicks and pops, and improper formatting.

There are a couple of ways to make your music louder without using extra outboard gear, though it may mean a software upgrade. Let's look at some ways that you can maximize the volume of each piece using your computer.

Normally, a song contains volume peaks that are at least 2 to 3 dB above the average signal level. (These peaks may not always seem obvious solely from listening, so you may want to check the file to see where the hottest signals are.) If the peaks are relatively low (-3 dB or so), consider normalizing the track. Normalizing brings the overall level up just enough that the loudest peaks hit a user-determined limit. You can bring the level up as high as -0.1 dB with some digital audio applications.

If, after normalizing, you still feel the need to raise the volume level, you may want to use a software compressor. This will allow you to increase the volume of the quieter parts, but it will keep the peaks at the limit you have set and thus avoid digital overload. If you still want to increase the volume level further, some CD-burning programs allow you to adjust the level of each track from a playlist in 1 dB increments. This is a little more dangerous because there is no safeguard that prevents a digital overload, so it's a good idea to use this last method conservatively.—Gino R.

ERROR LOG

August 1999, Letters, p. 12: The illustration was done by Paul Corio, not Barbara Pollack.

July 1999, Line 6 Pod review, p. 176: Although the review stated that the Pod's wah function is only accessible from the floor board, it can be controlled via MIDI using CC 43 for on/off and CC 4 for sweep.

June 1999, Opcode *Vision DSP* review, p. 154: The version we reviewed was actually 4.2, not 4.5. However, Opcode will most likely have released version 4.5 by the time you read this.

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

WHAT'S NEW

By the EM Staff

EGO-SYS WAMI RACK AND WAMI BOX

The WaMi Rack (Win; $699) from Ego-Sys combines audio and MIDI I/O and a SMPTE synchronizer in a package that includes a 1U rack-mount interface and PCI card. The interface has two analog inputs and eight analog outputs, all on balanced ¼-inch TRS connectors with 20-bit A/D and D/A converters. Stereo 24-bit S/PDIF I/O is provided on both coaxial RCA and optical Toslink connectors. The WaMi Rack supports sample rates of 32, 44.1, and 48 kHz. Eight 8-segment LED ladders monitor input or output levels.

The WaMi Rack has four MIDI In and four MIDI Out connectors, so the system can handle up to 64 MIDI channels. It also has word-clock I/O and SMPTE I/O and can sync to its internal clock, word clock, or other external clock sources. Audio and SMPTE control applications are provided. You'll need a Pentium PC with 32 MB of RAM and Windows 95 or 98 to operate the WaMi Rack.

Also new from Ego-Sys is the WaMi Box ($599; Win), a MIDI/audio I/O unit that has a PC Card interface for use with laptop PCs. The unit features two analog inputs and four analog outputs, all using 20-bit converters; S/PDIF I/O on optical and coaxial connectors; and one pair of MIDI In and Out connectors.

The WaMi Box contains 16 MB of sample RAM, a mic preamp, and a headphone amp. A RISC DSP chip provides internal mixing, reverb, chorus, parametric EQ, and delay. The unit supports Microsoft DirectSound. According to Ego-Sys, the WaMi Box has a signal-to-noise ratio of 108 dB and a frequency response of 20 Hz to 20 kHz. Thinkware (wholesale distributor); tel. (800) 369-6191 or (415) 777-9876; fax (415) 777-2972; e-mail sales@thinkware; Web www.egosys.net.

Circle #402 on Reader Service Card

ROLLS MB15 PROMATCH

The MB15 ProMatch ($80) from Rolls is a two-way signal converter that can take unbalanced stereo signals operating at -10 dBV and convert them to balanced +4 dBu signals. The unbalanced signal is on RCA jacks, and the balanced stereo signal is on XLR connectors. Another set of two balanced XLR and two unbalanced RCA connectors lets you perform the process in reverse.

There are two level knobs on the front of the device. By using RCA cables as "jumpers," connecting the MB15's outputs to its inputs, and routing balanced signals through the XLR ins and outs, you can also use the unit as a level controller for professional-level balanced signals.

The ProMatch provides -8 dB of gain using the +4 dBu inputs, and +20 on the -10 dBV inputs. Rolls rates the unit's noise floor at -88 dB and THD at 0.003%. Power is provided by a 9V adapter. Rolls Music Corp.; tel. (801) 263-9053; fax (801) 263-9068; e-mail eric@rolls.com; Web www.rolls.com.

Circle #401 on Reader Service Card

ROCKTRON CHAMELEON 2000

Rocktron's Chameleon 2000 ($719) is a 1U rack-mount digital guitar preamp that features the company's proprietary Digital Tube Replication Technology, which simulates the tone and response of tube-based guitar amps. The unit offers four main guitar tones, each modeled after a distinct vintage sound. "Clean American" replicates a Fender Tweed's clean response; "Texas Blues" is a Tweed sound turned to full volume. "Vintage British" is designed to respond in the manner of a Marshall tube amp, and "Mega Drive" turns up the gain for maximum crunch.

The Chameleon 2000 has 24-bit digital effects including delay, chorus, flanger, tremolo, compression, pitch shifting, and others. A simulator provides the sound of an amp run through a Variac for tube distortion at lower volumes. A programmable speaker-simulation section lets you use the Chameleon 2000 as a DI box. You can adjust parameters for speaker size, mic position, and reactance between an amp and a speaker load. Finally, Rocktron's patented HUSH noise reduction is employed at the input stage.

Front-panel knobs let you select presets and adjust effects parameters and input and output levels. A 5-segment LED input-level meter is provided. The unit has buttons for making A/B comparisons, bypassing the preamp and effects, and storing up to 254 programs. The delay, reverb, noise reduction, and speaker simulator are always available, and you can choose one additional effect. A one-line LCD shows the current tone and chosen effect, if any.

You can control every parameter on the Chameleon 2000 via MIDI. Mono, 20-bit audio input and output are provided on unbalanced ¼-inch connectors. Rocktron; tel. (248) 853-3055; fax (248) 853-5937; e-mail rocktron@eaglequest.com; Web www.rocktron.com.

Circle #403 on Reader Service Card

MAGIX MUSIC STUDIO PRO

Magix Entertainment has released Music Studio Professional (Win; $599.99), a suite of programs that includes *MIDI Studio* and *Audio Studio,* as well as CD-recording software and virtual mixers for both programs. The Professional package also includes an MPEG audio encoder that compresses audio into MPEG-Layer 2 format.

MIDI Studio is a digital audio sequencer that records an unlimited number of MIDI tracks at a resolution of 960 ppqn. Audio tracks can be recorded at 24-bit, 96 kHz resolution and saved as AIFF or WAV files. The program includes piano-roll, event-list, and drum-grid editors. There is a 3-band EQ and six effects returns in the mixer section; the effects returns can be used with up to three *MIDI Studio* effects and with DirectX plug-ins.

With the HyperDraw editor, you can control MIDI parameters such as Modulation, Velocity, and Pitch Bend. You can edit and print notation with the Score Editor; an Audio To Score function transforms monophonic audio signals into notation. *MIDI Studio*'s notation features accommodate 20 score styles, including drum notation for GM drums. Synchronization is provided via MTC or MIDI Clock.

Audio Studio is a 64-track hard-disk recording program that records 16- or 24-bit audio files at sampling rates of 22, 32, 44.1, 48, 88, or 96 kHz and enables you to burn Red Book CD-Rs. For each track, you can control the intensity of audio effects (such as reverb, chorus, and delay) by drawing a curve. *Audio Studio*'s mixer gives you three aux sends, onboard dynamics processing, delay effects, and support for DirectX plug-ins. Other available editing features include time stretching, pitch shifting, resampling, and normalizing.

Music Studio Professional runs on Pentium PCs with Windows 95/98, 64 MB of RAM, and a 16-bit sound card. Magix Entertainment; tel. (888) 866-2449 or (310) 656-0644; fax (310) 656-0234; e-mail info@magix.net; Web www.magix.net.

Circle #404 on Reader Service Card

TONEWORKS PANDORA PX-3

Toneworks' Pandora PX-3 ($250), the successor to the company's PX-2, is a pocket-size (4.6 x 0.9 x 2.8 inches) multi-effects processor. It is designed to provide guitarists and other instrumentalists with 56 effects, including reverb, pitch shifting, wah, overdrive, compression, and distortion. Other effects include a two-second digital delay and a cabinet resonator effect that simulates six speaker-cabinet configurations. There are 50 preset programs in all and 50 user-configurable locations. The Hyper Bass Boost button adds extra low-end punch to the signal.

New to the PX-3 is a phrase trainer that allows you to sample up to 16 seconds of digital audio and perform time expansion and compression by up to 25 percent of the original length. You can practice parts using the Pandora's metronome or one of 40 preset rhythm tracks. These tracks include both drum parts and bass lines; you can select any key for the bass part and use the PX-3's Intelligent Pitch-Shift function to create key-sensitive harmony lines.

By plugging a CD, MiniDisc, or cassette player into the PX-3's ¼-inch Aux In jack, you can play along with prerecorded tracks. A Key Transpose function can also be used to change the pitch of prerecorded tracks up or down 12 semitones.

The Pandora PX-3 is powered by four AAA batteries or a 9V adapter. Korg USA, Inc. (distributor); tel. (516) 333-9100; fax (516) 333-9108; e-mail product_support@korgusa.com; Web www.korg.com.

Circle #405 on Reader Service Card

APOGEE WYDE EYE A/D

Apogee Electronics has improved and expanded upon its Wyde Eye audio cables with the new Wyde Eye A/D line. Updates in design make the new cables highly suitable for both analog and digital applications.

The cables come in 110Ω balanced format, with a double shield of gold foil and braiding for clarity and RF rejection; 75Ω coaxial Wyde Eye A/Ds are also available. The new cables have distinctive temperature-resistant purple jackets that are marked to show the optimum signal-flow direction. The cables are designed to minimize noise and last a long time.

Wyde Eye A/D cables are available in various lengths, with assorted connector types—XLR, RCA, BNC, and ¼-inch TRS—to handle all the major digital and analog audio interfaces. Prices range from $30 for a 1.6-foot cable to $135 for a 32.8-foot cable. Apogee designed these cables to provide extremely low jitter for an improved dynamic range, and to deliver natural-sounding highs, smooth mids, and well-defined low end.

To help keep your studio tidy, Apogee includes a cable-tie with each Wyde Eye cable, which comes in a reusable zippered pouch for taking to gigs. Apogee Electronics; tel. (310) 915-1000; fax (310) 391-6262; e-mail info@apogeedigital.com; Web www.apogeedigital.com.

Circle #406 on Reader Service Card

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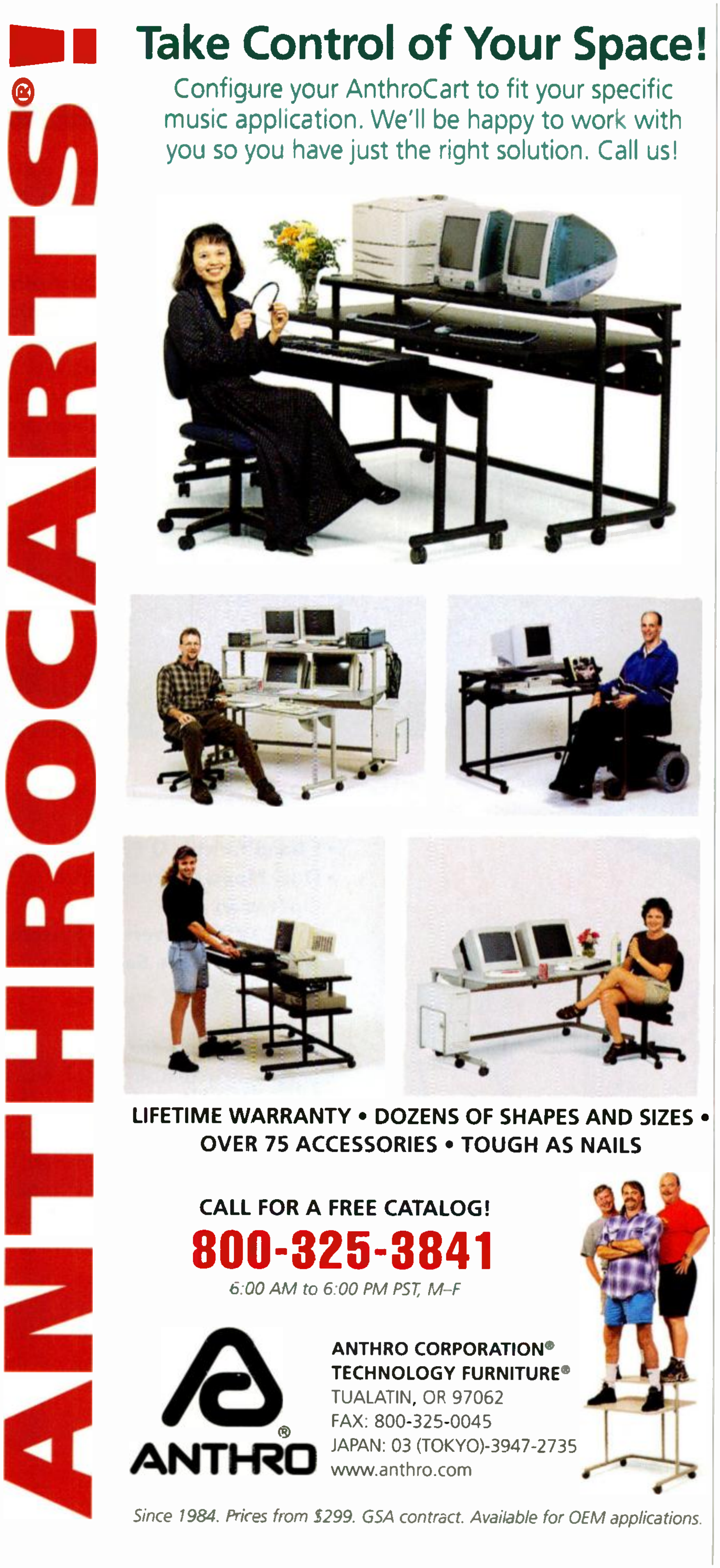


KEY CHANGES

New from **Metric Halo Labs** are two plug-in versions of the company's *SpectraFoo* audio-analysis software ($100 to $1,000, depending on features included). One version is written for the MOTU Audio System, the other for Pro Tools/Mix systems. Each version allows *SpectraFoo* to be used on up to 24 channels... Another plug-in newly ported to the MOTU Audio System is *Auto-Tune* ($399), from **Antares Audio Technologies.** The company has also released a less expensive VST version of the pitch-correction software. *Auto-Tune VST LE* ($99) differs from other versions only in that it lacks the Graphical Mode for detailed tweaking...**Sibelius Software** is shipping the Mac version of its *Sibelius* notation software ($599). This version is a completely Macintosh-native application, and it uses the same file format as the Windows version, so you can easily load *Sibelius* scores on either platform...**Symbolic Sound** has released a new version of its Kyma sound-design workstation hardware. The Capybara 320 Sound Computation Engine ($3,300) has 24-bit converters, four Motorola DSP56309 chips, and 96 MB of sample RAM. In addition, the unit features extensive synchronization options and the ability to add DSP chips, I/O, and sample RAM...**E-mu** has released version 4.0 of the EOS operating system for Emulator 4 samplers. Also newly available for the Emulator 4 are the ADAT I/O and DWAM cards ($549 and $395, respectively). The ADAT I/O lets you sample on two of its eight Lightpipe input channels and output 16 tracks of audio on two 8-channel streams. The DWAM card gives you MIDI, AES/EBU, and word-clock I/O, as well as an ASCII connection, so you can use your sampler with a PC-compatible keyboard...**TC Electronic** is now distributing and marketing Dynaudio Acoustics products.

—Rick Weldon

GEFEN SYSTEMS

On Gefen's *Sounds of the American West* four-CD set ($295), you'll find sounds of the frontier days. The first disc, *The Old West,* provides rumbling stagecoaches, eggs frying on the griddle, meat sizzling over the fire, prospectors panning for gold, and more. Other discs in the series include *Industrial Revolution; Trains, Guns, and Model Ts;* and *Western Town,* which has spittoon clangs, saloon door swings, and telegraph clatter.

Digiffects CDs ($60 each) are intended to serve as part of a general-purpose sound library. Two (of the ten) new offerings, *Stockholm* and *South Africa,* include ambient sounds recorded in those locales; *Industry, Human, Special, Cross Section, City,* and *South America* round out the ambient sounds in this selection. Two new *Zynthetic Harmony* discs are included; these discs, from the Production Elements product line, feature a variety of sounds designed to establish moods or punctuate audio material such as voice-overs.

Also new is the *Foley Sound Library* five-CD set ($395), a collection of sound effects by Academy Award–winning sound designers at Soundelux Hollywood. This set gives you everything from the sounds of brawling to sword fighting, shattering glass and splintering wood to fizzing soda pop. Gefen Systems (distributor); tel. (800) 545-6900 or (818) 884-6294; fax (818) 884-3108; e-mail gsinfo@gefen.com; Web www.gefen.com.

Circle #407 on Reader Service Card

PATCHMAN MUSIC

Patchman Music's latest Wind Controller Soundbank set ($39.95) can be used with MIDI wind and breath controllers and is compatible with Alesis QS-series synthesizers, including the QS6, QS6.1, QS7, QS7.1, QS8, QS8.1, and QSR models. The Soundbank contains 127 sounds that range from woodwind, brass, and string patches to synth leads, basses, guitars, a General MIDI drum kit, morphed sounds, and others.

The morphed patches are programmed to gradually change from one sound to the next, according to the force of your breath. Other patches are split, so you can use different sounds with a single patch, depending on what register you are playing in. There are string sounds that are dependent on Velocity; the sound switches from a bowed string sound to pizzicato when you tongue the note harder. Some of the more extreme sounds, including PhilGlass and Dimentia, rely on the QS synths' Modulation Matrix capabilities for mapping various modulation sources to multiple destinations.

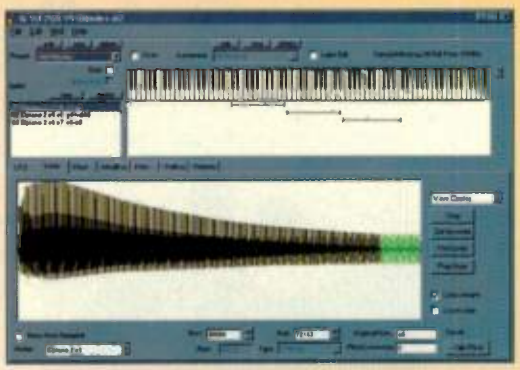
You can buy the Wind Controller Soundbank as PC-compatible SYX files, or in Standard MIDI File or Mac Self Loader formats. Patchman Music; tel. (216) 221-8282; e-mail matteblack@aol.com; Web www.patchmanmusic.com.

Circle #408 on Reader Service Card

MZONE

The *Wavelt Synth* CD-ROM (Win; $99), the fourth release in the Wavelt series from Mzone (formerly from Time Signature), is now available. *Wavelt Synth* contains a SoundFont library of synthesizer sounds sampled from a wide array of instruments, ranging from classic analog synthesizers to modern physical-modeling units, as well as a variety of drum machines. The sounds can be used with any SoundFont-compatible sound card or software synth.

The disc includes more than 300 SoundFont banks, each with numerous variations, for a total of more than 2,000 synth sounds. A large selection of synth drum sounds is available, both as complete kits and as banks of individual sounds. You also get looped sequences of drums, basses, synth leads, and vocoders to play with, in three tempi and several keys.



The *Wavelt Synth* package includes *Wien* (as in Vienna), a SoundFont editor that gives you a variety of synth parameters to tweak so that you can create your own patches. You can have fun with your custom patches and the preset ones using the included *MicroLogic Fun* sequencing program from Emagic. *MicroLogic Fun* features real-time MIDI parameter controls and editing tools and provides notation capability.

Wavelt Synth requires at least an 80486/66 PC with 16 MB of RAM and a SoundFont-compatible sound card or synth program. East West (distributor); tel. (800) 833-8339 or (212) 541-7221; fax (212) 541-7015; e-mail info@mzone.dk; Web www.mzone.dk.

Circle #409 on Reader Service Card

PERFORMANCE TOOLS ▲▲▲▲

▼ ELECTRO-HARMONIX

Electro-Harmonix has released the Q-Tron+ Envelope Follower ($278), an upgraded version of the company's Q-Tron. Designed by Mike Biegel, inventor of the classic Mutron III envelope filter, the Q-Tron+ uses an optical filter and has a knob that lets you choose between highpass, lowpass, or midrange-bandpass filters and Mix mode, which combines the bandpass-filtered and unprocessed sounds. A Drive switch selects the direction of the filter sweep.

You can use the unit's Range knob in its low position to emphasize vowel-like sounds, or in high position to bring out harmonic and other overtones. The Boost switch and gain knob work in tandem: the gain knob functions as both a volume control and a filter-sensitivity control in Boost mode, whereas in Normal mode it acts only as a filter control. The Peak Control knob determines the frequency peak of the selected filter.

New on the Q-Tron+ is an effects loop that lets you place a sound between the unit's envelope-detection circuitry and filter sections without changing the envelope drive. Using the effects loop, you can process a sound using the filter section without changing that sound's natural envelope. The Q-Tron+'s Attack Response control allows you to select between a slow attack and the faster attack of the original Q-Tron.

The rear panel features a single input and output, an effects send, and an effects return, each on an unbalanced ¼-inch connector. The unit has an external power supply and is hand assembled in the United States. Electro-Harmonix; tel. (212) 529-0466; fax (212) 529-0486; e-mail info@ehx.com; Web www.ehx .com.

Circle #410 on Reader Service Card

DIGITECH

DigiTech's new RP-21 ($849.95) is a combined tube preamp and multi-effects processor in a floor unit, suited for guitars or other instruments in live or studio applications. The unit has unbalanced ¼-inch analog outputs; stereo, S/PDIF digital outputs on RCA connectors; and MIDI In, Out, and Thru ports. The instrument input is on a balanced ¼-inch TRS jack.

The top of the box presents ten foot-controlled buttons for selecting effects; five of these are preset, and the other five are assignable. Two other buttons control Bank Up/Down, trigger the Bypass function, and access the built-in tuner. Above these are an LCD screen and controls for selecting and editing programs. On the right side is an expression pedal, which can control volume or other parameters.

The RP-21's 100 factory presets include tube-preamp tones and a multitude of digital effects; there are also 100 user locations for custom patches. Besides the basics—chorus, delay, flange, reverb, tremolo, wah, and so on—it comes with goodies such as reverse delay, mono and stereo samplers, and detuning. For example, a special reverse delay called Time Warp slows the delayed pattern to a complete stop and then starts it up again in reverse. The RP-21's distortion comes in solid-state and tube flavors, and these can be run in parallel and even split between the left/right channels.

DigiTech rates the RP-21's signal-to-noise ratio at >98 dB (A weighted) and total harmonic distortion at less than 0.03% (1 kHz). DigiTech; tel. (801) 566-8800; fax (801) 566-7005; e-mail customer@digitech.com; Web www .digitech.com.

Circle #411 on Reader Service Card

▼ TECH 21

Bass Compactor ($125), a new pedal for bass players, is out from Tech 21, makers of the SansAmp. This stompbox offers compression and active tone controls that can be used independently or in combination. According to the manufacturer, by applying the Bass Compactor's EQ section, you can make your passive bass sound active. You can also skip the EQ and just apply a little compression to rein in harsh peaks (handy for those of you who slap). Or, of course, you can use both.

The compression knob gives you a range of compression ratios from 1:1 to 15:1. The EQ section has high and low bands with ±12 dB cut/boost. Equalization can be either pre- or post-compression. A fourth control knob is for output level, providing up to 6 dB of clean boost.

The Bass Compactor runs on a 9-volt battery or an external DC power supply, which is available separately. Tech 21; tel. (212) 315-1116; fax (212) 315-0825; e-mail info@tech21nyc.com; Web www .tech21nyc.com.

Circle #412 on Reader Service Card

NOTEHEADS IGOR

Swedish company Noteheads boasts that its *Igor* notation program (Mac/Win; $545) has a drag-and-drop user interface that's easy enough for the complete novice to use, yet sophisticated enough to suit professional publishers, arrangers, and composers. As you input scores from a MIDI keyboard, or import them as Standard MIDI Files, the program analyzes each part and assigns appropriate dynamic and articulation markings.

Igor reproduces all markings during playback, with articulation styles specific to each instrument. It can automatically switch MIDI patches depending on articulation and can even crossfade between them. As you input parts, *Igor* will default to the proper clefs for the 260 instruments in its database, and it knows the range of each instrument. With sensitivity to the type of instrument, it can find appropriate points for page turns. *Igor* supports beams and tuplets across bar lines.

Page layout in *Igor* can be completely customized, including stem width, notehead size, thickness of bar lines, symbol definition, and more. The program will automatically avoid collisions in placing elements, and it will push staves apart as needed to accommodate markings or ledger lines. *Igor* can sort instruments in the correct order and spell their names horizontally or vertically.

An unusual feature of *Igor* is its support for notational devices used in early music; its instrument library includes many early instruments, such as rebec, hurdy-gurdy, and lute. For modern music, it can handle microtones and keyless music and offers four methods of notating accidentals.

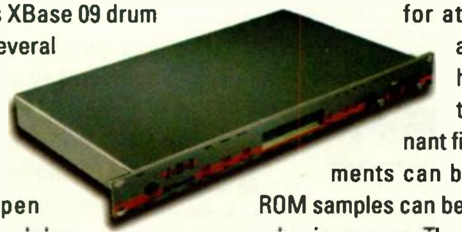
You can link separate files that are part of a single work. There is unlimited undo and redo, even after saving, with separate undo histories for each open score. *Igor* can read and write NIFF format files.

To run *Igor* on a Mac, you need a PowerPC 601 with Mac OS 8.0 and 20 MB of RAM. For PC, you need a Pentium processor with Windows 95/98/NT and 20 MB of RAM. Noteheads Musical Expert Systems; e-mail info@noteheads.com; Web www.noteheads.com.

Circle #416 on Reader Service Card

JOMOX AIRBASE 99

Jomox has released the Airbase 99 ($995), a 1U drum module that uses some of the sounds and technology behind the company's XBase 09 drum machine, along with several new instruments and features. Ten instruments are on the module: bass, snare, high and low tom, open and closed hi-hat, handclap, rim shot, crash, and ride cymbal.



Bass and snare sounds and parameters on the Airbase 99 are identical to those of the Jomox XBase 09. These sounds, along with the tom-toms, are analog; all others are ROM samples that are routed through a VCA modulated by an envelope generator with controls for attack, peak time, and decay. The hi-hat can be routed to a 2-pole resonant filter. All the instruments can be tuned, and the ROM samples can be played either forward or in reverse. There are two assignable LFOs that can be synched to MIDI.

On the front panel are a volume knob, value increment/decrement knob, 2-line LCD, MIDI indicator lights, and trigger button. Jomox supplies Airbase 99 editing environments for several digital audio sequencers. The module has 500 factory preset sounds in ROM, and there are 1,024 user programs.

On the rear panel are a discrete output on an unbalanced ¼-inch connector for each of the ten instruments; a pair of unbalanced ¼-inch jacks for a master stereo mix output; MIDI In, Out, and Thru connectors; and a ¼-inch TRS headphone jack. SoundBox (distributor); tel. (323) 769-5510; fax (626) 822-0110; e-mail soundboxla@aol.com; Web home.earthlink.net/~johnnyvn.

Circle #417 on Reader Service Card

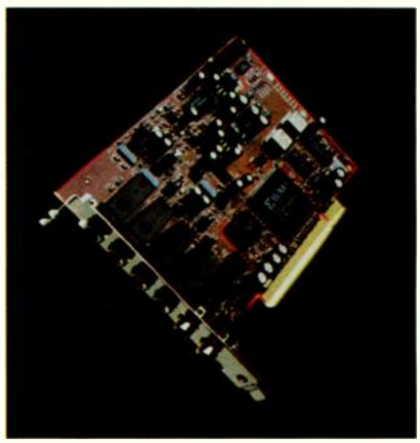
DAL CARDDELUXE

Digital Audio Labs is offering a new audio card for PCs that supplies analog and digital I/O. The CardDeluxe ($595) PCI card records and plays back at 8- to 24-bit resolution with sample rates from 8 to 96 kHz. DirectX drivers are available for Windows 95, 98, and NT; ASIO drivers should also be ready by the time you read this, and Mac support is planned.

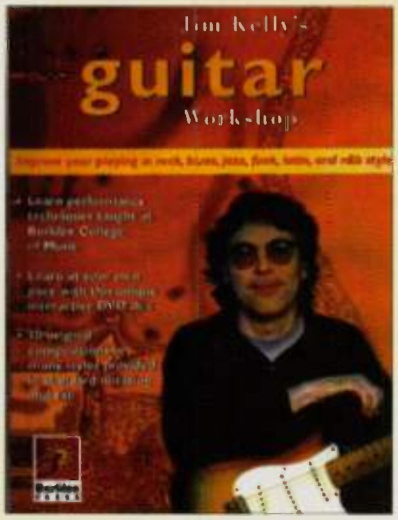
The CardDeluxe provides two channels of analog I/O on balanced ¼-inch TRS jacks and stereo S/PDIF digital I/O on RCA connectors. (An optional AES/EBU and S/PDIF optical interface is forthcoming.) You can run all four channels simultaneously, and you can link multiple CardDeluxes and slave them all to the same sample clock. DAL's WavSync feature ensures that all tracks will start recording or playback at the same time, even when you use multiple cards.

The CardDeluxe can operate at either +4 dBu or -10 dBV signal level. Frequency response is rated at 20 Hz to 20 kHz. Digital Audio Labs; tel. (612) 559-9098; fax (612) 559-0124; e-mail dalinfo@digitalaudio.com; Web www.digitalaudio.com.

Circle #418 on Reader Service Card



GET SMART! ▲▲▲▲



▲ BERKLEE PRESS

Berklee Press, the publishing division of the Berklee College of Music, has released its first interactive DVD disc. *Jim Kelly's Guitar Workshop* (Mac/Win; $29.95) is an instructional disc that provides full-screen, full-motion video and CD-quality audio. The disc is designed to provide intermediate and advanced guitarists with lessons that represent a cross-section of modern guitar styles, and song forms that reflect the sounds of guitarists such as Jeff Beck, Wes Montgomery, and Stevie Ray Vaughan. Multiple language tracks let you take the lessons in English, Spanish, or Japanese.

Using the DVD, you can choose what material to cover and thereby progress at your own rate. You can search for songs on the disc either by style or by title. Each song features video clips of Kelly, and you can view him playing from three angles: full body, left hand only, and right hand only. Any or all of these can be viewed at any time during playback.

Jim Kelly's Guitar Workshop can be used with DVD-compatible PCs or Macintoshes. Berklee Press; tel. (617) 747-2146; fax (617) 747-2149; e-mail berkleepress@berklee.edu; Web www.berkleepress.com.

Circle #413 on Reader Service Card

▼ ALFRED PUBLISHING

Alfred's *Essentials of Music Theory*, vol. 1, CD-ROM (Mac/Win; student version $29.95; educator version $99.95) offers lessons in basic theory with an emphasis on ear training. The program, which is based on the best-selling book series of the same name, uses demonstrations played on piano, trumpet, flute, clarinet, sax, and violin to demonstrate concepts aurally and includes musical examples from a variety of periods and cultures. Lessons are narrated to aid with pronunciation of new musical terms and to help younger students who might not yet read fluently.

Exercises reinforce each new concept and are ordered randomly, so students can replay exercise sections as needed for practice. Each unit of the course culminates with a review that is scored so that students can monitor their progress. The Educator version includes record-keeping capability for the scores of up to 200 students.

The program also includes a Glossary and Index of Terms and Symbols, which gives definitions of all terms and symbols covered in the lessons, with spoken pronunciations and aural examples to illustrate entries. Volumes 2 and 3 of the program are scheduled to be released next January.

The Macintosh version requires at least a 68040 processor running Mac OS 7.1, 16 MB of RAM, and a 4x CD-ROM drive. Minimum requirements for the PC are a 486-DX2 CPU running at 66 MHz, Windows 3.x or higher, 16 MB of RAM, a Windows-compatible sound card, and a 4x CD-ROM drive. Alfred Publishing; tel. (818) 891-5999; fax (818) 891-2182; Web www.alfredpub.com.

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▼ AKAI

Akai has released the GCF1 guitar chord finder ($19.95), a device the size of a business card that has a library of 324 guitar chords and 905 patterns. The GCF1's LCD can display more than 900 chords, with different voicings and chord inversions. These are shown in a guitar tablature–style grid that displays a 6-string, 4-fret area.

Using buttons on the face of the GCF1, you can search for specific chord, notes within a given key signature, chord inversions, and tensions. An on/off button is found on the GCF1's face, and the unit shuts off automatically after five minutes without use. The GCF1 requires two AA batteries. Akai Musical Instrument Corporation; tel. (800) 433-5627 or (817) 831-9203; fax (817) 222-1490; e-mail akaiusa@ix.netcom.com; Web www.akai.com/akaipro. ❁

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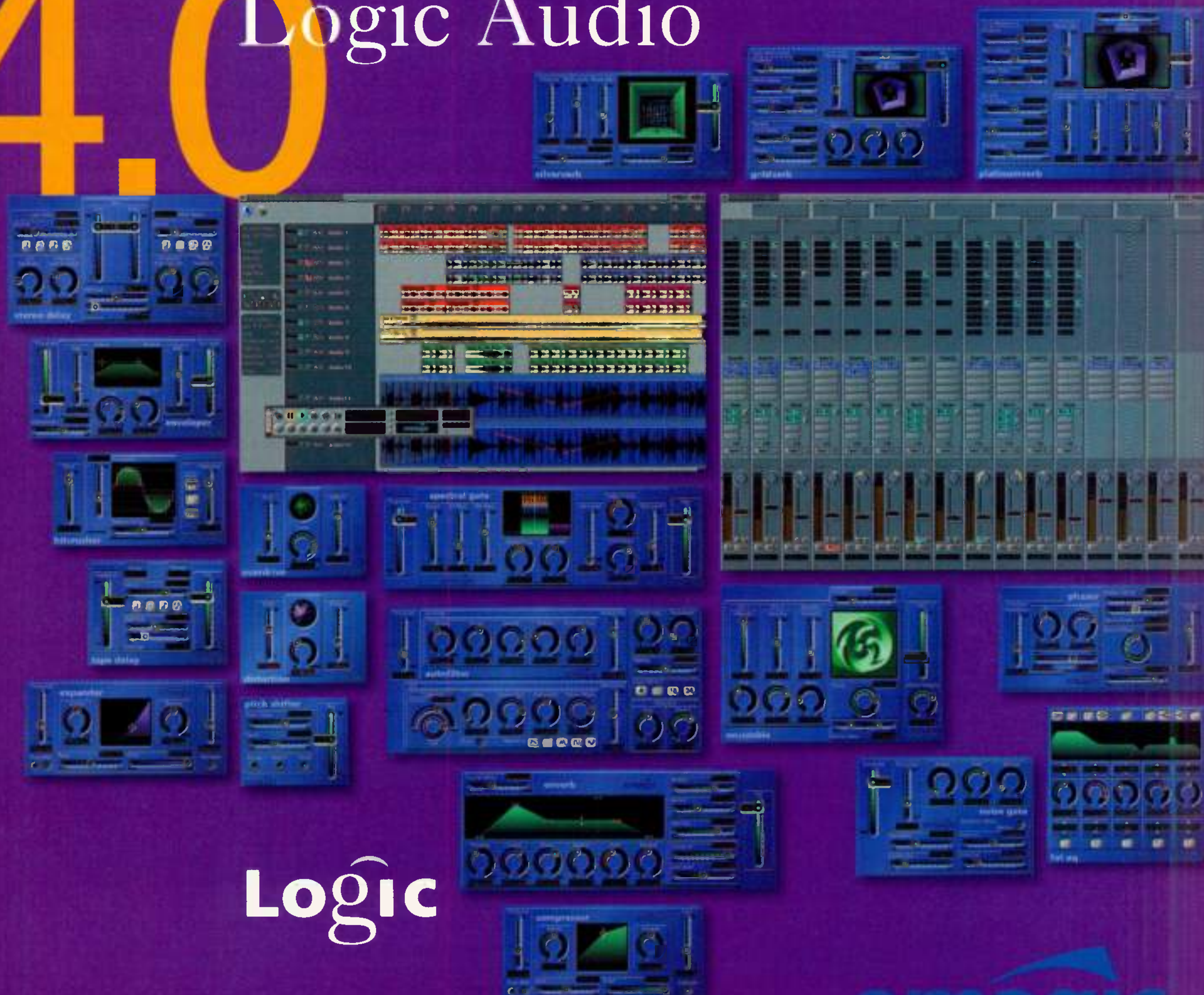

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Technology with Soul.

TECH PAGE

Microscopic Microphones

A new breed of microphones breaks the size barrier.

By Scott Wilkinson

The job of a sonic transducer is simple: to convert an audio signal from one form into another. Speakers convert electrical signals into acoustic sound waves by vibrating a diaphragm in response to an alternating current. Similarly, microphones convert acoustic sound waves into an electrical signal when a diaphragm vibrates in response to fluctuating air pressure.

The design of these transducers has not changed much since their development decades ago. For example, most microphones remain relatively large, bulky devices to house the associated electronics and protect the delicate diaphragm from damage. However, there are several new microphone designs that could change all that.

Among the most interesting approaches to microphone miniaturization is the Microflown. Invented by Hans-Elias de Bree in 1994, the Microflown does not measure fluctuating air pressure; instead, it measures the velocity of air particles across two tiny, resistive strips of platinum on silicon nitride, which are heated to about 200 degrees Celsius. (In fluid dynamics, the motion of gas or liquid particles is called *flow,* hence the name Microflown.)

When air flows across the strips, heat is carried from one to the other, causing a temperature differential between them. This results in a resistive differential, which generates a minuscule voltage proportional to the particle velocity. In the case of a sound wave, the air flow across the strips alternates according to the waveform, which results in a corresponding alternating voltage.

It's important to understand that an air particle is not a gas molecule but rather a volume of air that is small compared with the dimensions of the measuring device (which I'll discuss in a moment) or the sound's minimum wavelength (about half an inch at 20 kHz in air). The amplitude of the sound wave determines the particle velocity; for example, 94 dB SPL corresponds to a particle velocity of 2.5 millimeters per second, and 0 dB SPL (the threshold of hearing) corresponds to a particle velocity of 50 nanometers per second. The particle velocity can be increased at the sensors by placing carefully designed obstacles very nearby. These obstacles can form the structure in which the sensors are housed, which can increase the amplitude of the alternating flow by as much as 20 dB.

The entire Microflown (sensors and electrical connections) measures a mere 3 mm wide, 2 mm long, and 0.3 mm thick (see **Fig. 1**). The sensor strips are an amazing 0.0003 mm thick, making them impossible to see with the naked eye. In addition, the Microflown has no moving parts, which means it's highly reliable and exhibits no resonances. It's also more resistant to extreme ambient conditions, such as moisture and dirt.

To develop the Microflown and its various commercial applications, de Bree started Microflown Technologies (www.microflown.com) in 1998 with Alex Koers. Microflowns are fabricated in a clean-room environment using techniques similar to those found in semiconductor manufacturing. As a result, their performance characteristics fall within tight limits; for example, differences in sensitivity between individual units should be less than 1 dB, which is quite low compared with traditional microphones.

The Microflown is highly directional (the polar pattern is figure-8 all the way down to 0 Hz), and because it measures air velocity instead of pressure, it is very sensitive to near-field sources while effectively rejecting far-field sources. Its frequency response extends from 0 Hz to beyond 20 kHz, but its natural sensitivity drops by 6 dB/octave above 1 kHz. This can be corrected with the appropriate electronics, and a low-noise studio mic is planned for next year. With companies like Sennheiser and Brüel & Kjaer expressing serious interest in Microflowns, who knows what exciting products might emerge in the future?

COURTESY MICROFLOWN TECHNOLOGIES

FIG. 1: Two Microflowns (foreground) are tiny compared with a match and B&K's smallest conventional mic.

DANCE OF THE DEAD

BY – PETER – M^c^CONNELL

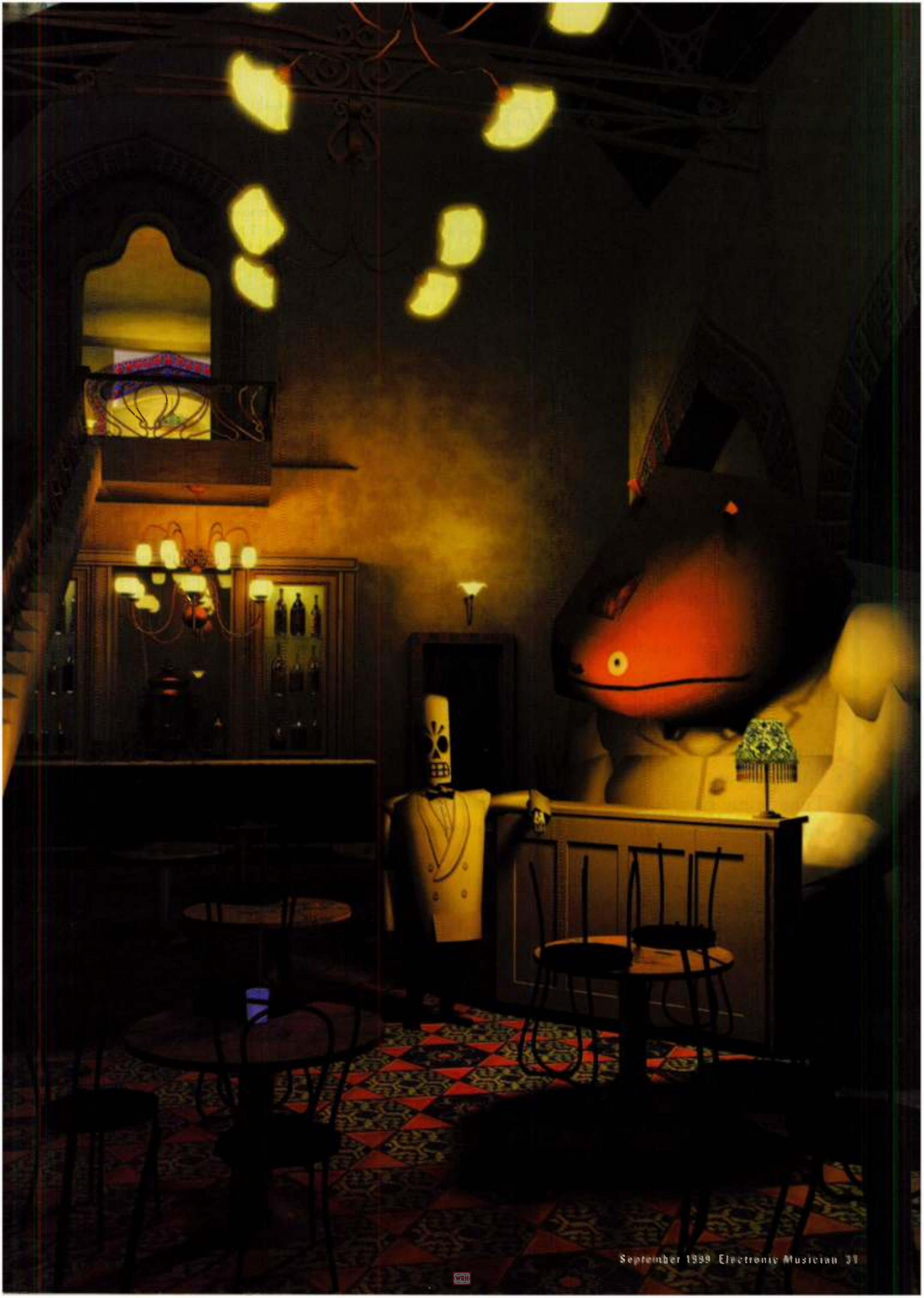
THE ADVENTURES OF A COMPOSER CREATING THE GAME MUSIC FOR GRIM FANDANGO

In addition to creativity, producing music for games requires a keen sense of balance in negotiating aesthetic decisions, budget constraints, and technical issues while working within a fast-paced and ever-shifting schedule. Now that PC and console game technologies allow for high-resolution music recorded live, the stakes have been raised higher than ever. Given the impact that live performance adds to a soundtrack, and the vast musical territory that must be covered with a limited budget and short schedule, key parts must be chosen and recorded with great efficiency. Every take must count, and never was this more true than when I worked on the score for LucasArts' *Grim Fandango*.

Grim Fandango is a graphic adventure game, which means that it's like watching a movie while you control the actions of the main character as he or she interacts with the environment and the other characters in the game. *Grim Fandango*'s leading man, Manny Calavera, sells travel packages to souls who journey in the Land of the Dead. But something is awry in this film-noir-like world. The travel tickets are going to souls who don't deserve them, leaving the good souls without a way to the Ninth Underworld, or Land of Eternal Rest.

One of these victims is the lovely Mercedes Colomar, for whom Manny has developed a more-than-professional interest. Manny, along with his demon driver and sidekick, Glottis, must find Mercedes and get her the tickets she deserves. Thus the scene is set for a four-year journey through a fantastic land with cities of sin, corrupt port towns full of casinos and shady characters, wild forests, an underwater slave city, and a majestic Mayan temple in the mountains. Bringing this diverse world to life with music was a huge task that yielded three hours of finished score—as much music as you would typically hear in three feature films.

Manny Calavera and his sidekick, Glottis, share a rare casual moment in Manny's café. In this scene, we hear Glottis noodling absent-mindedly on the piano. As Glottis leaves, muffled jazz can be heard from the casino next door. Such use of "source" music (rather than underscoring) was made more believable by employing live performers.

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DANCE OF THE DEAD

on a sort of virtual orchestra set up on an imaginary stage in such a way that there was a space for small ensembles of (mostly swing) musicians in front of the stage.

This setup was created using two E-mu E4s maxed out with 250 MB of memory, along with an E-mu Procussion and a Roland Sound Canvas. Good sample libraries were essential. I set up a template in MOTU *Digital Performer*, so that each virtual instrument was accessible on one of 68 MIDI channels (64 E4 channels and two channels each for the Sound Canvas and Procussion modules). External processing was provided by a Lexicon PCM-70 and an Alesis Q2.

With most of the major themes written and approved and a MIDI/synth configuration established, I was ready for the killer nine months of heavy production. At this point, it's worth taking a little detour to discuss the roles that music plays in an adventure game such as *Grim Fandango*. Each role has different implications with regard to live recording.

ON A ROLE

This type of adventure game has essentially three kinds of musical pieces: ambient pieces, event-triggered episodes, and underscoring for noninteractive movies (called "cut scenes" at LucasArts). In planning the production of these kinds of music, you must keep several things in mind: How does the player experience each piece in the game? How often will a piece be heard? Does the music need to loop or repeat, and if so, for how long? Ambient pieces provide the background for most of the action in the game. Event-triggered episodes and cut scenes, however, are likely to be experienced only once by the player and therefore may not be worth the time and expense of producing several live parts. On the other hand, these types of pieces serve as rewards to the player and showpieces for the game, so you must strike a balance.

The schedule further complicates the situation because cut scenes and one-time animations usually are not ready until late in the production process, and even then they're on an unpredictable timetable. This creates difficulties when scheduling recording sessions—writing the music in the allotted time is hard enough, let alone recording it. There were weeks during which I had to compose as much as two

Unlike the heroes of most games, *Grim Fandango*'s Manny Calavera is dead throughout the game's entirety. Part of the challenge in scoring this game was to evoke other-worldliness while keeping the score familiar sounding.

DANCE OF THE DEAD

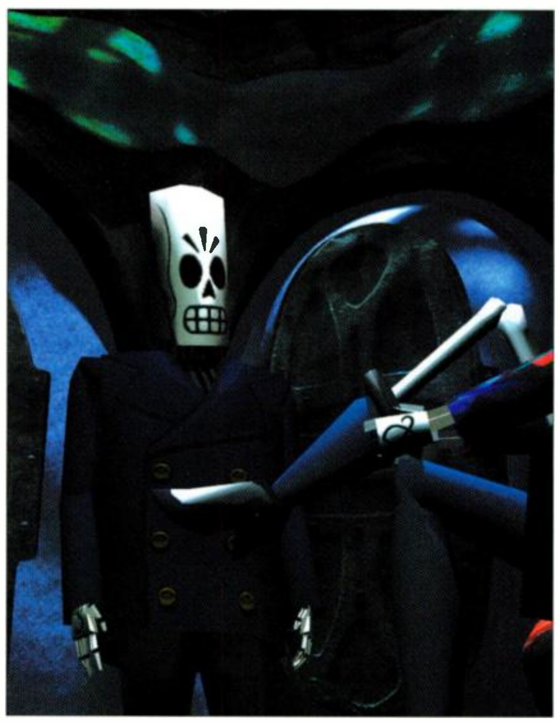
live parts directly into *Digital Performer* without tying up the control room. You could call this technique "guerrilla production": it's quick and dirty but allows you to knock off lots of live parts with minimal effort. Then when the main studio is really needed, you can bring it into the picture with only minor reconfiguration. To streamline the operation even more, we use Farallon's *Timbuktu* software to command the composer's machine from the control room. That way, there's no running back and forth. Everything can be done in the control room.

Recording in LucasArts' main studio is the most rewarding part of the production process. I get to take off my composer's hat, put on a producer's hat, and collaborate with the musicians and a great engineer to bring the score to life. Of course, it can be tense, too; the clock is always ticking, and the budget is always dwindling. This is where it helps to have an ace engineer—always a luxury in our industry. Jeff Kliment was the man behind the faders on *Grim Fandango*, and he made sure that everything sounded great and that the sessions ran smoothly. Kliment was also lead sound designer for the game, so he was responsible for the final mix of music, dialogue, and sound effects—in other words, the overall sound of the game.

Another important way to keep the recording process fun and efficient is to be clear about what you want as a producer and to discern what each musician can bring to the music. One player may be a phenomenal reader, for example; another may give you something amazing if you just provide a few melodic and stylistic guidelines and cut him or her loose. On *Grim Fandango,* I soon learned that with music based so heavily on swing jazz, getting the best results often meant not being too attached to the written score—no matter how good I thought my original ideas were.

I took this approach to a new extreme as the recording sessions progressed, and it actually changed my compositional technique. As explained earlier, I often have to schedule sessions in groups, before the music is written. I realized that live playing was really giving life to the score. In fact, the playing mattered almost more than the notes themselves. What's more, I could benefit doubly by taking advantage of improvised sections of music to cover more ground in the game than I could if I painstakingly recorded multitracked parts. In other words, I could hit more musical targets in less time.

With this in mind, Kliment and I started recording musicians in twos and threes so that they could play off each other. We then used digital editing in order to shape these "wild" sessions into more structured pieces. In some cases, I even had one of the musicians, Hans Christian (a producer in his own right), do this editing himself. Then I added synth orchestra



A single sexy trumpet line was all it took to capture the mood of this scene, in which Manny reacts to Mercedes Colomar changing her nylons.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parts to fill out the textures and provide more harmonic context. That was exactly opposite from the order in which we did the more "standard" pieces in the score. The result was a truly alive feeling borne of the spontaneity and collaborative nature of the recording process—a huge benefit to the overall musical score.

The mixing session usually follows the recording sessions. In this case, however, we had to be superefficient, so we merged the two stages by mixing some pieces before others were recorded. Everything about scoring for interactive media revolves around one basic fact: you don't know how much you will get to finish until you're near the end. You can't record three hours of multitrack music, mix it down, then dump it in one big truckload on the project, and expect it all to be programmed into the game during the week before final testing. Instead you have to mix early and often, to get the flow of finished pieces coming into the project as soon as possible.

Remember that for the engineer, mixing involves not just the music but the integration of the ambient sound-effects bed, which is a major part of the final sound. In the case of cut scenes, there are also dialog tracks and foreground sound effects. And there is also an important phase in which levels are adjusted so that the cut scenes and mixed ambient pieces all match each other.

Finally, the entire soundtrack is slowly scrutinized in one last meticulous pass that takes a couple of weeks. During this time, the engineer tweaks the levels of all the sonic elements until they balance each other and the music and sound programming work the way they should. This process alone is fodder for another complete article; suffice it to say that, because it accounts for perhaps a third of the perceived overall sound quality, we take it very seriously.

FINAL TAKE

As Clint Eastwood once said, "A man's gotta know his limitations." Every project has its constraints. Indeed, I believe that good art thrives under constraints. While working on a big interactive project like *Grim Fandango*, it's imperative that you keep your sense of perspective. You want the greatest part of the experience to have the best quality possible. That always means cutting some corners somewhere. Looking back, for example, I wish I had been able to find a better piano sample, but I'm not sorry that I didn't record the piano live. Too many other things would have suffered.

Unlike Manny Calavera and his friends, the art and craft of scoring to interactive picture exist in the real world. So does the payoff—playing a finished game with a rich sonic landscape, and knowing that thousands of other adventurers can do the same.

When **Peter McConnell** *isn't composing and editing music at LucasArts Entertainment, he sings and plays electric violin in the Bay Area band Spinray.*

Encompassing the heritage of more than 20 years of delivering professional tools to the professional industry, TC Electronic proudly introduces a high-end vocal processing tool aimed at reducing tedious engineering time spent on doing vocal re-takes - dedicated to the professional vocal recording engineer.

The Intonator not only provides the ultimate solution to vocal pitch correction, but offers various highly useful tools as well, including adjustable De-esser and Adaptive Lo-Cut (ALC™) filtering techniques.

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THERE IS NO QUESTION THAT DIRECT RECORDING WITH GOOD GEAR WILL GIVE YOU THE CLEANEST SOUND. And no instrument reflects the efficiency of direct recording more than the human voice. However, the average vocal signal takes a trip through various gain stages in the mixer's channel strip, through the send/returns to effects (such as compression and EQ), and down the mix bus before hitting the recorder. Electrically speaking, that adds up to quite a distance and can translate into added noise and signal degradation.

For avoiding such noise and degradation while still getting the effects essential to recording vocals, there are handy devices on the market that reduce the number of parts the signal goes through and thus minimize the length of the path from input to output. These devices are often referred to as channel strips and voice processors.

The Path of Least

A RECORDIST'S GUIDE TO THE WORLD OF VOICE PROCESSORS.

By Gino Robair

Resistance

Photography by Philip Kaake

ONE VOICE, ONE CHANNEL

The terms *channel strip* and *voice processor* are often used interchangeably. That's because both kinds of units have a similar purpose: to get the signal to the recorder as directly as possible. But the two types also have significant differences.

Stand-alone channel strips are designed to take the place of a mixer's channel strip. The stand-alone has the same features as the channel strip on a mixer: mic preamp, line-level input, shelving filters, and parametric EQ. Besides the shorter signal path, the big difference is that the stand-alone version contains better components than those in the average personal-studio mixer. When you spend $600 per channel, you get components comparable to those in a high-end mixing desk.

Voice processors are also designed for direct recording and contain the same features as a channel strip, but with a few additions. Engineers regularly use dynamics processors when recording vocals because of the voice's wide dynamic range. That means you will want a compressor, limiter, expander, gate, and de-esser in the unit. Of course, these effects work well on other instruments, too, such as electric guitar, bass, and keyboards.

The layout of a voice processor should be as intuitive as that of a channel strip. In a mixer, the signal generally flows from top to bottom. The signal flow in a voice processor is similar, except that you lay the unit on its side so that the signal runs from left to right, entering the input stage at the far left, going through the various effects stages, and ending up at the output on the far right.

Don't limit yourself by using voice processors only for tracking. These units can prove equally beneficial during mixdown or for fattening prerecorded tracks. For this reason, voice processors have a variety of input and output options to maximize their usefulness in the studio.

CHOICES AND MORE CHOICES

Manufacturers all have different ideas as to what their direct-recording device should include and how the features should be organized. Because of the enormous number of possibilities, I have limited myself to discussing a handful of products for the purpose of comparison.

Voice processors can range in price from around $500 to $4,000. I decided to look at single-channel units priced between $630 and $850 that combine a mic preamp, compressor/limiter, expander/gate, and EQ. Within these parameters, I looked at a host of other useful features (for example, tube emulation, digital converters, and independently accessible effects), and chose a collection of processors that complemented, rather than fully duplicated, each other.

Besides the five that I have chosen to cover here, there are many other voice processors in this $630-to-$850 range. Therefore, as I discuss specific features, I will also briefly point to some of these others as examples. But before we examine our five contenders, let's review the various features that make up a voice processor.

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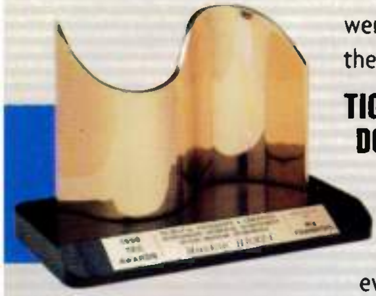
WHY DOES THE RESPONSE OF BECAUSE IT'S

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.

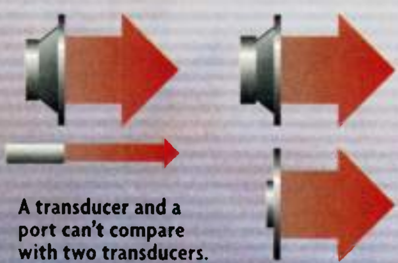


Last fall we won the pro audio industry's coveted TEC Award for best near field monitor. Modesty prevents us from listing the impressive field of competitors but you'll probably encounter their ads in this magazine.

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.

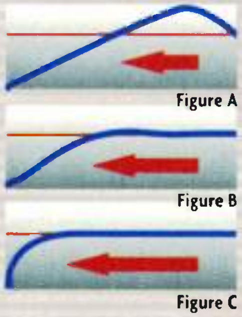
ANOTHER TRANSDUCER INSTEAD OF A PORT.

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.

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.

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

FIG. 2: With separate I/O in each section, the dbx 1086 allows you to use the dynamics processor and mic preamp independently. Included here is the 504X digital output card.

INS AND OUTS

When you're looking for the ideal direct-recording solution that is also useful for mixing, the more I/O possibilities you have, the better. In addition to a mic input, there should be line- and instrument-level inputs. The ability to tap in and out of the individual effects is a plus. At the other end, having balanced (+4 dBV, ¼-inch TRS or XLR) and unbalanced (-10 dBu, ¼-inch TS) line-level outputs is essential. Let's take a closer look at each section.

Mic preamp. The microphone preamp is one of the most important parts of a voice processor because it sets the stage (literally) for your sound. In order to record with the best signal-to-noise ratio possible, the output level of the microphone (-30 to -60 dBu) should be brought up to line level (-10 dBu or +4 dBV).

Usually, this change in level will color the sound somewhat, and in many cases that's good. One joy of engineering is tone sculpting, and the mic preamp can have as much to do with this sculpting as the microphone itself. For example, a tube preamp will have a different character than a solid-state unit. In addition, Class A circuitry (solid-state or tube) will give you superior sound and performance but will often cost a little extra.

Ultimately, the degree of transparency or coloration that you want the preamp to impart depends on your personal taste and the demands of the music. Other considerations include the recording medium that you're using. Engineers with tapeless studios may want a preamp that adds "tube warmth" or can emulate the effects of tape saturation. On the other hand, users of analog tape may want a transparent sound, with as little coloration as possible. The call is a completely subjective one. What sounds harsh and buzzy to one engineer may sound warm and fuzzy to another.

Another consideration is how your collection of mics will sound through the preamp: invariably, some mics will sound better than others. There are no rules that say you can't run a cheap mic through an expensive preamp or vice versa—which preamp you choose should be determined by whatever works best for the style of music that you're recording.

Examples of voice processors with tube preamps include A.R.T.'s Pro Channel ($799), Manley's Voxbox ($4,000), and Avalon Design's VT-737SP ($2,295). Solid-state designs include the Focusrite Voicebox MKII ($1,345) and the Rane VP12 ($599). The Manley, Avalon, and Focusrite are more expensive because they use Class A circuitry.

A mic preamp can include a couple of features that make life easier. First, it should have +48 volt phantom power so that you can power the mics that need it. A phase-reversal switch is also useful for changing the phase of the signal by 180 degrees. In addition, preamps often have a -10 or -20 dB pad to attenuate high-output mics and a low-cut filter to remove low-frequency energy such as rumble. Most voice processors on the market include many or all of these features.

Instrument input/DI. If you have ever recorded electric guitar or bass, you know how handy it is to have a compressor, gate, and EQ at your disposal. Having access to a tube stage is another plus. You'll probably want to plug your instrument into your voice processor and take advantage of these effects. Fortunately, many of these units have a ¼-inch, instrument-level input jack. The instrument input (or DI) raises the signal from instrument level (-30 to -20 dB) to line level. Look for processors with convenient ¼-inch front-panel jacks.

FIG. 1: The dbx 1086 Mic Pre Processor includes a mic preamp with a Detail feature, de-esser, expander/gate, compressor, and a PeakStopPlus limiter.

Line-level input and output. In addition to using a voice processor on mics and guitars, you can use it on keyboards, sound modules, and prerecorded tracks during mixdown. Manufacturers often provide both +4 dBV balanced and -10 dBu unbalanced line-level inputs to facilitate this type of use.

Voice processors also typically have a -10 dBu and a +4 dBV output. Depending on your studio situation, having the option of both output levels means that the device will have upward compatibility if you're moving from so-called semipro gear (which typically operates at -10) to professional gear (operating at +4).

Compressor/limiter. The dynamic range of the voice goes beyond what the average recorder can handle, so engineers routinely use compression when tracking vocals. The type of compressor and how well it operates can impact how the signal sounds after the fact. Consequently, take great care when choosing and using a compressor.

With a dedicated compressor you will usually have control over threshold level, attack and release times, ratio selection, and make-up gain, as well as the choice between hard or soft knee (that is, how suddenly compression begins).

Limiting is at the extreme end of compression, when the ratio is 10:1 and beyond. Limiting effectively inhibits any increase in output level no matter how far the signal goes above the threshold.

Compressors and limiters have three common designs: photo-optical, VCA, and variable mu. Photo-optical compressors (also called photoelectric or just plain "opto") differ in several ways from those compressors governed by a voltage-controlled amplifier (VCA). Opto compressors use a photosensitive resistor controlled by light-emitting diodes that are triggered by the input signal. Opto units compress over a narrower range (with less extreme compression ratios) and impart a livelier sound to the signal. Because of their simple user interface, they lend themselves to tweaking by ear. Examples of photo-optical compressors include

A.R.T.'s Tube Channel ($499), Avalon Design's VT-737SP, and Focusrite's Platinum VoiceMaster ($749).

VCA units, on the other hand, are far more predictable, are easier to use, have greater ratios, and allow for more flexibility and precision with attack and release times. Voice processors with VCA compressors include the MindPrint En-Voice ($749), PreSonus VXP ($700), and Focusrite Red 7 ($2,995). The En-Voice compressor includes a tube saturation stage.

Variable mu compressors (*mu* in this case means "gain") use a vacuum tube as a variable resistor to control the signal level. Manley uses a combination of variable mu and optical compression/limiting in its Voxbox. The A.R.T. Pro Channel gives you the choice of either opto/tube or variable-mu gain control.

Expanders and gates. Downward expansion and gating are processes used to reduce noise. Expanders create a greater dynamic range by progressively attenuating sounds that fall below a specified threshold; that is, they make the quiet sounds quieter. You can think of an expander as the opposite of a compressor, which brings up the level of the quieter sounds.

A full-featured expander gives you control over the threshold where the processing begins, the ratio of expansion, and the speed of the processing. Expanders are used primarily for eliminating extraneous sounds such as bleed from headphones, lip smacks, and clothing noises. Usually used during mixing rather than recording, the expander is often placed after the compressor to reduce the low-level noise that compression brings out. Many voice processors, however, locate the expander *before* the compressor, giving you the opportunity to reduce noise before the compression stage.

At the extreme end of downward expansion is gating. Rather than simply reducing the signal level when it drops below the threshold, a gate shuts the level off completely. Gating is used as much to shape the envelope of signals as to remove unwanted noise. For example, gates are used to fatten up drum sounds by quickly attenuating the instrument's natural decay. You can also use a gate to keep the buzz of an amp from passing through the unit when a guitarist isn't playing. Typical gate parameters include threshold and rate.

EQ and filters. Direct control over the timbral aspects of a signal is always welcome. Whether you use EQ to enhance the signal, remove unwanted anomalies, or repair damage caused by compression, you will want a band or two of parametric EQ as well as high- and low-shelving filters.

A parametric EQ lets you control three essential parameters: the exact frequency that you want to affect; the bandwidth, or Q, which is the range of frequencies around the center frequency; and volume for cutting or boosting.

Shelving filters allow you to modify the extreme ends of the signal and are usually found in the input section of voice processors. The most common is the highpass filter (also called a low-cut filter) which attenuates everything below a specific frequency. This is especially useful for removing rumble and mic-handling noise. Because the human voice doesn't go below 80 Hz, a typical cutoff frequency for a highpass filter is 75 Hz. Some units give you variable control over the cut frequency, sometimes ranging from 15 to 320 Hz. At the other end of the spectrum is the lowpass (or high-cut) filter, which affects everything above a specific high frequency, such as hiss.

For those of us with a limited number of mics, parametric EQ supplies the tools with which to expand the sonic palette. EQ is also a must when tracking instruments such as the electric guitar. Ranges in which the guitar can often use help include the lower-mid range, around 500 Hz; between 3 to 6 kHz for added bite; and from 8 to 10 kHz for sparkle. Unlike a graphic EQ, a parametric EQ enables you to locate the

FIG. 3: The Drawmer MX60 includes a handy front-panel ¼-inch input for instrument-level signals, and a 3-band saturation stage called Tubesound.

Voice Processors Specifications

	dbx 1086	Drawmer MX60	Focusrite Platinum VoiceMaster	HHB Radius 40	LA Audio PS-1
Price	$750	$629	$749	$749	$850
Inputs	(2) XLR; (1) ¼" TRS	(1) XLR; (1) ¼" TRS; (2) ¼" TS	(1) XLR; (1) ¼" TRS	(2) XLR; (2) ¼" TS	(1) XLR; (1) ¼" TRS; (4) ¼" TS
Outputs	(2) XLR; (2) ¼" TRS	(1) XLR; (1) ¼" TRS; (1) ¼" TS	(2) XLR; (1) ¼" TS	(1) XLR; (1) ¼" TS	(1) XLR; (3) ¼" TS
Frequency Response	<5 Hz to >180 kHz (-3 dB)	16Hz–28 kHz (-1 dB)	10 Hz–200 kHz (-1 dB)	10 Hz–40 kHz (-1 dB)	20 Hz–20 kHz (±1 dB)
Signal to Noise	Mic EIN: -122 dB	Mic EIN: -128 dB	Mic EIN: < -134 dB	Mic EIN: -127 dB	Mic EIN: < -128 dBu
Dynamic Range	116 dB	107 dB	116 dB	106 dB	112 dB
Dimensions	1U x 9" (D)	1U x 7.9" (D)	1U x 10" (D)	2U x 7.9" (D)	1U x 7.9" (D)
Weight	7 lbs., 3 oz.	7 lbs.	11 lbs.	5 lbs., 8 oz.	6 lbs., 10 oz.

FIG. 4: With the Drawmer MX60, you can simultaneously use balanced and unbalanced outputs and balanced and unbalanced inputs.

exact frequencies that need attention.

De-esser. A de-esser provides a form of frequency-dependent compression used to remove sibilance when recording vocals. Sibilants, such as *s* and *sh* sounds, have a high-frequency concentration between 3 and 8 kHz. Traditional de-essing is done by splitting the vocal signal, sending one side through the compressor's audio input, and the other side to an EQ patched into the compressor's sidechain input. The exact sibilant frequencies are boosted on the EQ, which makes the compressor more sensitive to them. The EQ-enhanced signal causes the compressor to attenuate the sibilants in the direct signal. Meanwhile, the signal going into the sidechain is not heard.

A de-esser increases the usability of a voice processor. However, a poorly implemented de-esser can sound more like a lowpass filter than a frequency-dependent compressor.

Insert and sidechain. As with a mixing console, an insert lets you introduce an additional outboard processor into the signal path. It also gives you an extra output point before the main output stage.

The sidechain input, a common item on dedicated dynamics processors, lets you control the compressor with an external signal. This is useful for ducking and frequency-dependent compression.

THE CONTENDERS

In designing a product such as a voice processor, manufacturers make presumptions about how recording engineers like to work and the kinds of features that they need most. For the sake of contrast and comparison, I have selected five units that have as many of the above features as possible, with just as many implementations.

dbx 1086. The dbx 1086 Mic Pre Processor ($750) stands out in the crowd in several ways. The most interesting for the personal-studio owner is that the mic preamp and the dynamics processor can be used independently. This allows you to simultaneously track a vocal while processing another signal. The other great feature is that the 1086 has room for an optional A/D converter, the dbx 504X. However, the 1086 is the only unit of the group that lacks a true parametric EQ section (see **Fig. 1**).

The 1086's preamp section has an XLR input and an XLR and ¼-inch TRS output. The dynamics section has XLR and ¼-inch TRS line-level inputs and outputs. Both output sections have a switch allowing you to choose between -10 dBV and +4 dBu output levels (see **Fig. 2**).

The preamp section has a variable-frequency low-cut filter (30 to 300 Hz)

FIG. 6: The Platinum VoiceMaster gives you the option of using a line-level output before the de-esser stage.

and a 2-band additive filter called Detail. When switched in, the Low control simultaneously boosts 125 Hz and cuts 400 Hz at a 2:1 ratio. The High knob adds up to 15 dB of the "air band" above 10 kHz. Having a variable-frequency Low Cut next to the Detail section baffled me initially. But I could imagine a scenario in which you would want to boost 125 Hz while cutting 70 Hz at the same time, and this unit will give you that option.

The 1086 has a single bypass switch for the entire dynamics section. However, the controls for the expander/gate, compressor/limiter, and de-esser each have an off position that effectively removes them from the signal chain. All of the buttons on the 1086 are backlit so that you can easily see their status, and the rotary controls use 40-position stepped, rather than continuously variable, pots.

The expander/gate has variable threshold and ratio controls, the compressor has threshold, ratio, and output-gain controls, and the de-esser lets you set threshold and frequency. The VCA compressor includes an OverEasy switch for soft- and hard-knee processing, and a Slow button that changes the speed of the compressor. The 1086 also includes the dbx PeakStopPlus two-stage limiter to keep the output from overloading the recorder input.

Drawmer MX60. Officially called the Front End One, the MX60 ($629) has all the traits of a well-equipped voice processor (see **Fig. 3**). The rear panel includes +4 dBu balanced and -10 dBV unbalanced ¼-inch line inputs, a mic input, and an insert jack. An instrument-level input is strategically placed on the front panel. The MX60 can simultaneously handle balanced and unbalanced line-level inputs, and balanced and unbalanced line-level outputs. Together, these features allow you to use the MX60 for level conversion (see **Fig. 4**).

The dynamics section of the MX60 includes a gate, compressor/limiter, and de-esser. The design of the MX60's VCA compressor is based on the DL241 and MX30. The three bands of EQ include a fully parametric mid band (with frequency, bandwidth, and cut/boost controls), as well as high- and low-shelving filters. Next is Tubesound, which has three separate adjustable bands of saturation: Lo covers 350 Hz and below; Mid handles 350 Hz to 2 kHz; and Hi is 2 kHz on up. Each band has a Drive control that ranges from 0 to 11.

The MX60 has a fixed-threshold (+20 dBu) prefade limiter before the master output fader. The limiter isn't user selectable—soft limiting begins automatically at +6 dBu—so you really have to watch how you manage the various gain stages to avoid unwanted limiting.

Finally, the MX60 has no power switch. To avoid sending damaging current spikes to your mics when you turn on your power strip or line conditioner, the unit's phantom power comes up and decays slowly.

Focusrite Platinum VoiceMaster. The Platinum series is Focusrite's foray into the price realm of the average personal studio. While products in their Red series cost upward of $2,500, the units in the Platinum line are priced well below $1,000 and contain many well-thought-out features based on the more expensive models.

For example, the Platinum VoiceMaster ($749) contains a Class A discrete transistor mic preamp with a frequency response of 10 Hz to 200 kHz (with -1 dB variance), which exceeds the human hearing range. Another notable feature is an opto compressor that includes a Treble control for reintroducing high frequencies into heavily compressed signals (see **Fig. 5**). The threshold and release times have variable controls, while speed and ratio are set with two-position buttons: Attack time is set using the Fast button, while Hard Ratio gives you a choice between high (6:1) and low (2:1) ratios. In addition, the VoiceMaster has an opto de-esser as well as a tunable saturation stage. An expander/gate and EQ section round out the feature set.

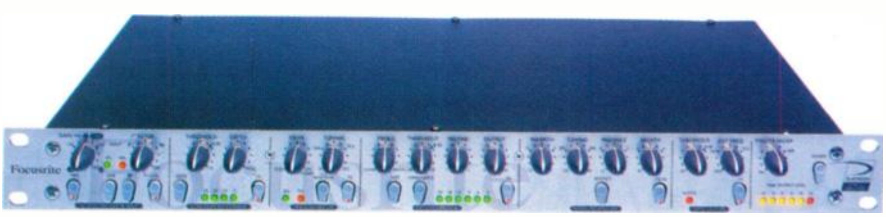


FIG. 5: Focusrite's Platinum VoiceMaster contains a Class A mic preamp, an opto compressor, and an opto de-esser.

VoiceMaster's back panel includes mic- and line-level inputs, an insert jack, and an XLR output that lets you take a signal out before it gets to the de-esser (see **Fig. 6**).

HHB Radius 40. HHB distributes TL Audio's Ivory Series 5051 Valve Processor in North America under the name Radius 40 Tube Voice Processor. (See the review of the 5051 in the December 1998 issue of **EM**.) Designed primarily for vocals, the Radius 40 ($749) is the only unit in this group without a built-in de-esser. A sidechain jack is included on the back panel so that you can de-ess the old-fashioned way.

Inside the Radius 40 are three tubes. The first is used for the input stage as part of a solid-state/tube hybrid circuit. The second and third are used in the compressor and 4-band parametric EQ stages. As the levels through the tube stages increase, the amber Drive LED illuminates. The red Peak LED signals that there is less than 5 dB of headroom left.

The input section includes a front-panel ¼-inch jack that can accept instrument-level and line-level inputs, and a switchable 90 Hz low-cut filter (see **Fig. 7**). The back panel has a mic input, ¼-inch unbalanced and XLR line-level inputs, the sidechain insert, a link jack for synchronizing the VCAs of two Radius 40s, an input-level switch, and XLR and ¼-inch unbalanced outputs (see

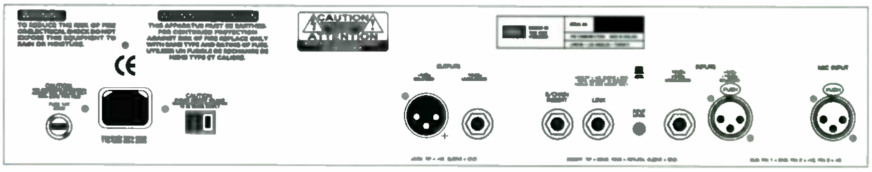


FIG. 8: The back panel of the Radius 40 includes both a link jack and a combination sidechain/insert jack.

Fig. 8). Interestingly, the Radius 40 lacks a phase switch, which seems an odd omission for a device that can be linked into a stereo configuration.

The Radius 40 has a proprietary solid-state compressor called a transconductance amplifier, with four fixed attack and release times. Threshold, ratio, and gain have continuously variable controls. The attack ranges from 0.5 to 40 milliseconds, and the release times are from 40 ms to 4 seconds. The compressor can be switched out of the signal path using a front-panel button.

Each of the EQ bands has four fixed bandwidths and a variable level control. The two mid bands have fairly wide bandwidths so that frequencies in adjacent bands overlap. The EQ section follows the compressor in the signal path but can be easily switched ahead of the compressor using a button on the front panel.

LA Audio PS-1. The most expensive voice processor of the bunch, the PS-1 ($850) has a mic preamp (with high- and low-cut filters), a full-featured compressor (including variable control over attack and release times), a parametric EQ with two variable mid bands, an expander, and a de-esser (see **Fig. 9**). In addition, the PS-1 can be fitted with an optional A/D converter.

The back panel gives you a good deal of flexibility by having separate inputs and outputs on the EQ and dynamics sections (see **Fig. 10**). The input section has mic, line, and DI inputs and a ¼-inch output. The dynamics processor has a line-level input and output as well as stereo-link and sidechain jacks. The EQ has line-level input and output, and the Output stage has a ¼-inch input next to the XLR output. All this I/O flexibility means that you can shorten the path to the recorder by using the preamp's ¼-inch line output, use the different effects independently, or reorder the effects in the signal chain.

SONIC FINDINGS

The best way to test a voice processor is on a voice. I asked singer/songwriter/guitarist Jill Garellick of Cactus Motel to be my test subject because of her dynamic singing and strumming style. I also recruited engineer Myles Boisen for his ears and gear. Garellick's guitar and voice were recorded separately using a matched pair of mics so that I could use two voice processors per take. I recorded the guitar using a factory-matched pair of Oktava MC012 microphones and tracked the voice with a matched pair of '70s-era Neumann U 87s. Each mic went through a single processor and then directly to tape.

To further test the behavior of the units, I ran a few different instruments through them, including guitar, bass, theremin, keyboards, and drum machine. I also routed tracks from tapes (music and spoken word) through them to see how each device reacted to various tracking and mixing conditions.

Pump up the volume. Each of the mic preamps in the collection has phantom power and a low-cut switch, and all but the Radius 40 have phase reversal. Other than features, what really sets these preamps apart from each other is how they sound; when heard side by side the differences are remarkable.

The combination tube/solid-state design of the Radius 40 mic preamp gives it more coloration than all but one of the other preamps, but with an added throatiness. Boisen described its sound as "warmer or murkier, depending on your taste," a quality that is no doubt due to the tube influence. The Radius 40 does, however, have a pronounced upper midrange that helps counteract its preamp's slight muddiness.

Heard on its own, the Radius 40 preamp is noticeably fuzzy. Even in the clean settings, the contours of the voice sounded like they were wrapped in gauze. Once the Radius 40 tracks were placed in the mix, however, they sounded great. The Radius 40 guitar track blended well with the vocals tracked through the other processors, and vice versa. With that in mind, I'd be less conservative with the degree of processing next time I use the preamp. Although it doesn't have as much "clean" headroom as the other preamps, you can push the overall level much further.

In contrast to the Radius 40, the PS-1's preamp had a smoother high end and an impressive clarity. The high harmonics of the guitar really came through. As far as tonal coloration, its preamp was neither the most nor the least transparent of the five units. Compared with the MX60 and the VoiceMaster, I detected a slight buzziness around the voice, similar to that of the Radius 40 but far less prominent. And the PS-1 preamp seemed to have quite a bit more headroom.

The preamp in the dbx 1086 was a little less satisfying to use. The stepped input gain was problematic at times: there were drastic changes with each step, and sometimes the level I wanted was between the notches. Adjustments to the level often affected the behavior of the compressor, making it more difficult to predict the response. And similar to the Radius 40, the 1086 had the least amount of headroom and required a greater input gain to get a level on tape that matched the PS-1, VoiceMaster, and MX60.

The 1086 was the least transparent preamp of the bunch. Hearing it on its own, I could detect a little coloration. In

FIG. 7: The Radius 40's front-panel ¼-inch input can handle instrument-level and line-level signals.

THE PATH OF LEAST **Resistance**

side-by-side comparisons, however, it sounded somewhat two-dimensional, and the coloration was far more pronounced. But coloration is not necessarily a bad thing. **EM** associate editor Brian Knave used the 1086 preamp to smooth out an otherwise harsh vocal sound. One might say that the 1086's preamp has a "soft focus" effect on the voice.

On the other end of the spectrum is the preamp in the MX60. Both the voice and the acoustic guitar sounded fantastic through it. Boisen noted a slight boost to the low mids, which

Voice Processors Features

	dbx 1086	Drawmer MX60	Focusrite Platinum VoiceMaster	HHB Radius 40	LA Audio PS-1
Mic Pre Type	solid state	solid state/Class A	solid state/Class A	tube/solid state	solid state
Low Cut	30–300 Hz	100 Hz	20–300 Hz	90 Hz	75 Hz
High Cut	n/a	n/a	n/a	n/a	12 kHz
Saturation Stage	no	3-band Lo, Mid, Hi	wideband and 1.4–7.2 kHz	no	no
DI/Front Panel	no/no	yes/yes	no/no	yes/yes	yes/no
Stereo Link	no	no	no	yes	yes
Phase Reverse	yes	yes	yes	no	yes
Sidechain/Insert	no/no	no/yes	no/yes	yes/yes	yes/no
Digital I/O	yes	no	no	no	yes
Sampling Rates	44.1/48 kHz	n/a	n/a	n/a	44.1/48 kHz
Resolution	16/20/24 bit	n/a	n/a	n/a	16/24 bit
Word Clock	in/out	n/a	n/a	n/a	in
Audio In	no	n/a	n/a	n/a	yes
Connections	S/PDIF; AES/EBU	n/a	n/a	n/a	S/PDIF; AES/EBU, Toslink
COMPRESSOR/LIMITER					
Type	VCA	VCA	photo-optical	proprietary/TCA	VCA
Threshold	-40 to +20 dBu	-40 to +20 dBu	-26 to +10 dBu	-20 to +20 dBu	-30 to +20 dBu
Ratio	1:1 to ∞:1	1.2:1 to ∞:1	2:1 and 6:1	1.5:1 to 30:1	1:1 to 20:1
Attack Time	fast/slow	auto	fast/slow	0.5–40 ms	1–100 ms
Release Time	fast/slow	auto	fast/slow	0.04–4 sec.	0.04–4 sec.
Make-Up Gain	-20 to +20 dB	-30 to +20 dB	0 to +20 dB	0 to +20 dB	-20 to +20 dB
EXPANDER/GATE					
Threshold	off to +15 dB	off to +20 dB	-40 to +10 dB	off to -20 dB	-70 to 0 dB
Expander Ratio	1:1.2 to 1:8	n/a	1:1 to 1:3	n/a	1:3
Attack Time	<100 µsec	auto	n/a	n/a	n/a
Release Time	auto	fast/slow	n/a	n/a	fast/slow
EQUALIZER					
Low Shelving	125 Hz, +15 dB	100 Hz, ±18 dB	120–600 Hz, -12/+8 dB	60–500 Hz, -12/+8 dB	80 Hz, ±15 dB
LM Band	400 Hz, -7.5 dB	150 Hz–16kHz, ± 18 dB	1.5 kHz, -12/+8 dB	250 Hz–2.2 kHz, -12/+8 dB	60 Hz–2.4 kHz, ±15 dB
HM Band	n/a	n/a	4.5 kHz, -6 dB	1.5–5 kHz, -12/+8 dB	500 Hz–20 kHz, ±15 dB
High Shelving	10 kHz, +15 dB	4.25 kHz, ± 18 dB	10 kHz, ±8 dB	2.2–12 kHz, -12/+8 dB	12 kHz, ±15 dB
Bandwidth	fixed	variable	fixed	fixed	variable
DE ESSER					
Frequency	800 Hz–10 kHz	male/female	2.2– 9.2 kHz	n/a	800 Hz–8 kHz
Depth	30 dB	off to 10 dB	-20 to +10 dB	n/a	0 to -20 dB

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added a bit of lumpiness to the sound. You have to mind the MX60's input level carefully, otherwise the prefade limiter starts working on the sound. If you want to avoid that stage altogether, take the +4 dBu signal out of the balanced ¼-inch insert jack on the back panel. The MX60 Class A preamp includes a phase reverse switch, a 100 Hz low-cut filter, a Brightness button, and a -20 dB pad.

Of the five mic preamps, the VoiceMaster is the most transparent. As the other Class A preamp in the collection, it's easy to see why. The high end is crisp and clear, and the frequencies are more evenly distributed than in the other preamps. I compared the VoiceMaster to the Focusrite Green Series Dual Mic Pre, and the sound was remarkably similar.

Total transparency may not always work for a session, so it's nice to have the ability to gradually color the sound to taste. Both the VoiceMaster and the MX60 give you a tunable saturation stage that, on the voice, sounded convincing in very small doses. And while we're discussing saturation, coloration, and buzziness, let's see what happens when we run line- and instrument-level instruments through these devices.

Instrumental connections. Because TL Audio/HHB's and Drawmer's designers were nice enough to put instrument jacks on the front panels, I'll begin with the Radius 40 and the MX60. The ¼-inch jack on the front panel of the Radius 40 can handle just about any signal. That's great, because going behind your rack each time you want to plug in an instrument is inconvenient. Keep in mind that it's an unbalanced input: if your keyboards are across the room and you need the RFI protection of a balanced line, you may end up going behind your rack after all. Be sure to have a TRS-to-XLR cable as well because the Radius 40's balanced input uses an XLR jack.

Like the Radius 40, the MX60 has an unbalanced ¼-inch input on the face, though it is meant for instrument-level signals only. The back panel has +4 dBu balanced and -10 dBV unbalanced ¼-inch input jacks, and you can use both of them, as well as the balanced and unbalanced outputs, at the same time.

In its back-panel input section, the PS-1 has jacks for line- and instrument-level signals, or you can go directly into the dynamics processor, equalizer, and output section if you like. The 1086 and the VoiceMaster do not have a built in DI. The only line-level input on the 1086 goes directly into the dynamics processor.

Dynamic differences. There are bound to be compromises when you jam a multitude of effects into one box priced under $850. This is most noticeable in the feature sets of the dynamics processors. As far as getting a full-featured compressor, the PS-1 and Radius 40 come the closest. They both offer control over attack and release times, as well as ratio, threshold, and make-up gain.

Besides having variable threshold (-30 to +20 dB) and ratio (1:1 to 20:1), the PS-1 is the only one of the bunch with continuously variable attack and release controls. In addition, it has a Gain knob with 20 dB cut or boost, as well as a hard/soft–knee button and a bypass switch. The PS-1 compressor can be used independently of the other stages, and it has a link function for stereo use.

However, the PS-1 compressor is more challenging to use than the other compressors in this group. The controls are very touchy, and the slightest movement noticeably affects the sound. And to get the meter bar to match what I was hearing, I had to push the input gain more than with the other units.

On the other hand, the dbx meters tell you the story right away. Although the 1086 has no release-time control, it is smoother sounding and more musical than the PS-1. You adjust attack and release times automatically on the 1086. By engaging the Slow button, you can extend the attack and release times for instrumental applications. But having so little control over the speed made the 1086 compressor challenging to use; getting the right setting was sometimes difficult due to the stepped controls.

Setting up the VoiceMaster's opto compressor, by contrast, was quick and easy. It has only two ratio settings to choose from—Soft Ratio (2:1) is intended for vocals and Hard Ratio (6:1) for instruments. However, the soft setting was perfect for acoustic guitar as well as voice. And having a variable release time helped in getting the right sound for electric guitar and bass tracks. The Treble feature proved useful in reinstating high frequencies post-compression.

The VCA compressor in the MX60 was a tad smoother than the VoiceMaster's opto compressor. Although the attack and release times are automatically controlled, I had no trouble getting a natural and musical sound. Besides a variable threshold control, the MX60 has a continuously variable ratio. The MX60's compressor cannot be linked for stereo use.

The Radius 40 has one of the smoothest, most gentle compressors of the group. It was easy to dial in a setting on both the guitar and voice, using medium-slow speeds and a ratio of about 8:1. Drum samples sounded especially beefy through this compressor. And having the option of switching the EQ in front of the compressor helped even out the compressor's response to bass-heavy signals.

Mind expansion. All but one of the voice processors in this group place the expander/gate *before* the compressor. The Radius 40's expander/gate is at the very end of the signal chain and is by far the simplest of the bunch. It is controlled by a single knob with a range of off/-50 to -20 dB. This expander/gate acts as a gate in the higher settings (fully clockwise, between -30 and -20 dB). However, this results in envelopes getting sped up and sharp transients being chopped—so much so that, when set at -25, it gave drum samples a reverse-sounding attack.

Although the Radius 40's expander/gate control comes after the master output control in the signal chain, the knob is located next to the make-up gain in the middle of the front panel.

FIG. 9: The LA Audio PS-1 includes a full-featured compressor and parametric equalizer.

This clued me in about how to use it. When I really pushed the input and make-up gain stages, I could use the expander/gate to keep the noise of the twice-boosted signal from coming through. Still, I missed having separate control over the speed of the gate. Other than this one application, I had a difficult time using this particular expander/gate successfully.

Compared with the Radius 40, the 1086's expander/gate is not only easy to set up but sounds fantastic. Threshold and ratio are your only controls, and with the 1086, that's enough to shape percussion and keep noise to a minimum. The threshold range is off/-80 to +15 dB, and the ratios span 1:2.1 to 1:8 (although they are mistakenly printed as 1.2:1 and 8:1 on the front panel). At times, the gate seemed a little touchy, probably due to its automated program sensitivity.

LA Audio refers to the PS-1's expander as Noise Reduction. This may seem slightly confusing at first, but as a downward expander, it delivers on its promise. The Noise Reduction function has a preset ratio equivalent to 3:1, a threshold control that ranges from -70 to 0 dB, and a two-position, switchable release time (fast or slow). The attack is preset with a soft-knee envelope that sounded natural on everything I ran through it. An amber light lets you know when the signal is above threshold. The back-panel link connector allows you to coordinate the expanders of two PS-1s for stereo operation.

The PS-1's knobs are highly sensitive, requiring very subtle movements to get the right timing. However, I had no trouble dialing in a natural decay each time. As with the PS-1's de-esser and compressor, you can use the expander independently of the EQ and input section. The three dynamics processors share the same I/O, so you can't plug into each of them independently.

The gate Drawmer put in the MX60 is actually more of a downward expander. The MX60's gate has two speeds (fast and slow) and a variable threshold (off to +20 dB). As with the other gates,

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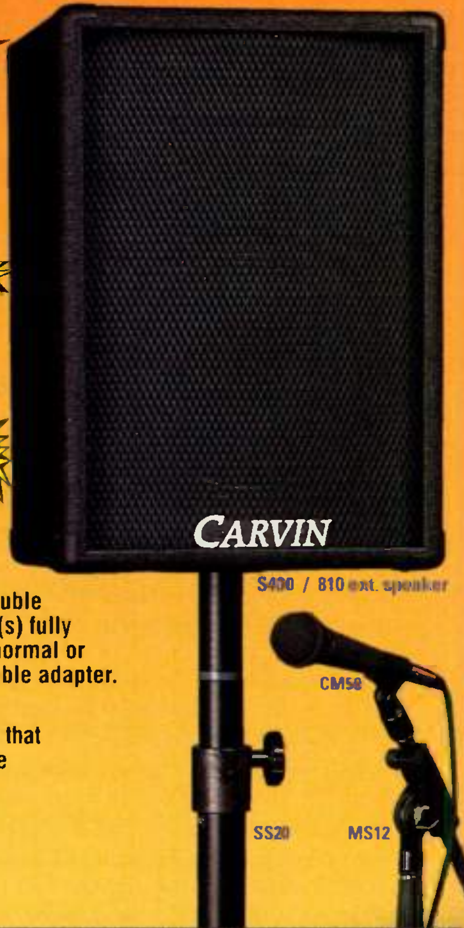

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changes in threshold will vary the speed: when you dial toward a higher threshold the gate moves quickly, and lower threshold settings (the MX60 descends to -70 dB) slow the gate down.

It was often difficult to make the MX60's gate sound transparent. However, when I did find the right setting for an application, it worked well. The combination of a low threshold with the slow speed setting was perfect for fattening a noisy drum machine. In this particular instance, the threshold control made it fast and easy to find a decay that sounded musical.

The VoiceMaster's Noise Reducing Expander includes a switch to choose either gating or expansion. The variable controls include Threshold (-40 to +10 dB) and Depth (0 to Full). A 4-LED bar graph indicates the amount of reduction being applied to the signal. The VoiceMaster's expander worked well with vocal tracks (both spoken and sung). But when the gate was engaged, the speed was so fast that it was difficult to find a good setting using only threshold and depth controls. This particular gate could use a speed control.

EQ review. Like dynamics processing, a parametric equalizer is handy for both tracking and mixing. The most useful parameters in the midrange frequencies are adjustable bandwidth, sweepable frequency range, and cut/boost control.

The PS-1 possesses the most dramatic-sounding EQ of the bunch, with two full-featured mid bands in addition to high and low shelving. Each of the PS-1's mid bands gives you control over a range spanning 60 Hz to 20 kHz. The MX60 has a similarly wide frequency range of 150 Hz to 16 kHz. However, this range is packed into a single parametric band with shelving filters on either side.

The EQ in both of these units worked so well that I wanted to use them on everything! And it was easy to zero in on the frequency I needed. These EQs are so accurate that you don't need much boosting or cutting to hear a change in the sound.

The EQ on the Radius 40 was far less dramatic, but no less musical than the PS-1 or MX60. The combination of wide bandwidths and overlapping frequencies made this EQ particularly useful. Having a tube stage in the EQ is also a plus if you want to add extra coloration.

Although the 1086 has no EQ section, the preamp has a variable-frequency low-cut filter. I didn't find the dual-band Detail feature particularly useful on voice or acoustic guitar, though it may prove useful with other miked instruments. I was disappointed that the 1086 had no line-level input before the dynamics processor: I suspect that Detail might work well on thin-sounding electronic instruments, but there's no way to try it.

The voice-optimized EQ on the Voice Master has five controls, cryptically named Warmth, Tuning, Presence, Absence, and Breath. This section worked well once I figured out how each of the controls works. Breath is a 10 kHz shelving filter for cutting or boosting the "air" frequencies. Tuning and

Warmth, combined, are like a low-frequency parametric EQ. Tuning determines the center frequency (between 120 and 600 Hz) that Warmth cuts or boosts. Presence is a fixed bandwidth cut/boost control with a peak at 1.5 kHz. The Absence button has a center frequency of 4.5 kHz and is intended to further attenuate the midrange by 6 dB. The area around 4.5 kHz is where the voice can sound harsh.

On the units that have equalizers, the EQ is placed after the compressor in the signal chain. Sometimes, however, you may want the equalizer *before* the compressor. For example, if a specific frequency is causing the compressor to pump, you can use the EQ to cut the offending frequency. Two of our voice processors give you this power: the Radius 40 has a front-panel button that lets you move the EQ ahead of the compressor, and the PS-1 allows you to repatch the order on the back panel.

Mind your s's and z's. The de-esser comes after the compressor on the 1086 and the VoiceMaster, and before the compressor on the MX60 and the PS-1. In fact, on the PS-1, the de-esser is the first stage after the input. On the MX60, the de-esser comes between the gate and the compressor.

The advantage of having the de-esser before the compressor is that you can remove sibilant peaks that cause unwanted compression. On the other hand, having the de-esser last in the chain is useful for removing increased sibilant frequencies caused by extreme compression. Unfortunately, none of the units in the group allow you to switch the order of the de-esser and compressor in the signal chain.

The de-esser on the 1086 uses a variable-frequency highpass filter to attenuate the sibilant frequencies. Consequently, it acted the most like a traditional sidechain de-esser. At the extreme settings I could hear the ducking action on sharp transients, but in normal usage this de-esser was more useful than those in the other processors. The 1086 de-esser followed the dynamic contour of the voice nicely, and at times it seemed to reach out and grab the sibilants. The 1086 also has a bar graph that visually indicates (from -1 to -30 dB) the amount of de-essing being applied to the signal.

The 1086 has a threshold and frequency control for its de-esser. The threshold is numbered from 1 to 10, and the frequency ranges from 800 Hz to 10 kHz. Having the option of going below the normal sibilant frequencies is useful for instrumental applications.

The PS-1 de-esser seemed less drastic but worked well, perhaps because of the sharp cutoff of the filter. The PS-1's de-esser uses a lowpass filter, sweepable from 800 Hz to 8 kHz. The Listen function gave this unit an advantage over the others. When Listen is active, you hear only what's above the filter, making it easier to locate the exact sibilant frequency. Boisen suggested that the precision of the PS-1 de-esser would make it harder to make a mistake with this effect.

The VoiceMaster's opto de-esser was the most subtle of the group. In fact, it's not a true de-esser at all, but rather a lowpass filter with a frequency range from 2.2 kHz to 9.2 kHz. One interesting feature of the VoiceMaster is that it includes an aux output that lets you tap the signal before it reaches the de-esser. That way you can use a de-essed signal for effects (such as a reverb), while sending the non-de-essed signal to tape. Because the VoiceMaster's de-esser is last in the signal chain, both signals will have identical amounts of dynamic and EQ effects.

The MX60 had the most easily audible de-essing effect of the group. Although Drawmer says the work is done by "intelligent circuitry," it sounds more like a shelving filter than a true de-esser, and it didn't take much attenuation to begin coloring the sound. Instead of having a variable frequency control, you choose between Male and Female tonalities. I found that the MX60's style of de-essing worked better on spoken-word material rather than on singing.

Tube or not tube. The MX60 and the VoiceMaster have features that emulate the "warm" sound of tape saturation and tubes. On the VoiceMaster, the saturation stage is before the compressor, whereas on the MX60 it's after the compressor and the EQ. A little goes a long way with each unit.

The VoiceMaster's Vocal Saturator creates smooth-sounding distortion at lower gain levels. Crank up the saturation level, and you get a delicious overdrive without a lot of noise or fuzz. The VoiceMaster gives you the option of tuning the saturation. With the Full Bandwidth button in, you get a warm, wide-band distortion. With Full Bandwidth out, you can use the Tuning control to precisely select the upper frequencies (between 1.4 and 7.2 kHz) you want emphasized. The Drive knob ranges from Clean to Unclean. I had to keep the Vocal Saturator button engaged, even when I didn't intend to use the effect. When everything but the Saturator was in the signal path, the VoiceMaster's self-noise became more noticeable. Once I added the Saturator into the signal path (with the level set to Clean), the hum went away.

The MX60 can be pushed much further than the VoiceMaster, and with the help of the 3-band Tubesound, I was able to tune in an amazing layer of distortion. It also helps that there is an extra gain control before the Tubesound stage. Having three frequency bands to work with made it much easier to get results that sounded convincing on voice. But then again, Tubesound sounded great on everything I put through it.

The most over-the-top sound came from the Radius 40 and its three tubes. With the input and compressor gains nearly maxed, and the EQ sent pre-compressor, I could fine-tune four frequency bands of distortion, which was enough to make the thinnest keyboard sound gigantic. You won't use this effect every day, but it's nice to know you have it.

Direct to digital. What could be handier than having digital converters built into your processor? The dbx 1086 and the LA Audio PS-1 both have optional A/D converters, though neither of the units I reviewed had the converters installed.

The dbx 1086 includes a space on the


FIG. 10: Each section of the PS-1 has its own input and output, so you can use the sections independently or repatch the signal flow.

rear panel for the new 504X Digital Output card ($400). This card uses the proprietary dbx Type IV A/D converters for 16-, 20-, and 24-bit resolution. The 504X includes both AES/EBU and S/PDIF connectors, with a switch to select the format. Additional buttons allow you to select 16- or 20-bit word lengths and 44.1 or 48 kHz sample rates. The 504X also has word-clock input and output, so you can slave the unit or use it as the master clock.

The 1086's front panel has a three-position Dither switch for the converter card. In the Off position, the unit sends a 24-bit signal through the digital output. The two types of dither are TPDR and SNR². There is also a Shape button to select Type 1 or Type 2 noise shaping.

The optional A/D converter for the PS-1, the PS-DR ($329.95), is also 24-bit capable. It has word-clock input only, but it adds a Toslink optical output to the S/PDIF and AES/EBU connectors. Like the 504X, the PS-1 can run 44.1 and 48 kHz sampling rates, but you can dither only to 16 bits. The PS-1's digital card includes an additional ¼-inch audio input, so you can run two processors through one digital card. Keep in mind that it must be installed at the factory; you may be better off buying the PS-1D ($1,149), which has the card preinstalled.

These two units aren't the only voice processors in this price range that have digital capabilities. The MindPrint En-Voice uses the DI-Mod 24/48 card ($249), which has digital input *and* output capabilities via S/PDIF connectors. This means that you can use the DI-Mod to run a track from a digital recorder through the En-Voice's tube compressor and back to the digital recorder. The DI-Mod also includes an extra ¼-inch input so that you can use both digital channels of the S/PDIF connector.

BALANCING ACT

As with any piece of gear you use, a voice processor must match your style of working. Deciding on which features you need most will help you pick the

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processor that's right for you. If you prefer quick-and-easy results over flexibility, you'll want a unit with one or two buttons per effect. If you like to tweak, however, then features and knobs may beckon you.

HHB's Radius 40 is the easiest to use of the five voice processors. The compressor and EQ set up quickly, and the tubes give them a nice sound. The tube/solid-state hybrid mic preamp gives the voice a bit of an edge that helps it sit well in the mix. Although I didn't find the Radius 40's expander/gate that useful, and it didn't have a de-esser, I love that you can easily put the EQ before the compressor. It may be called a tube voice processor, but it works wonders on instruments, as well.

The dbx 1086 has several big things going for it: you can use the mic preamp and dynamics processor separately; the expander/gate and de-esser are fantastic; and the optional 504X A/D converter allows you to go direct to digital. The mic preamp had the most coloration of the group, but the competition in this area was stiff. The dynamics processor is easily the high point of the 1086.

The LA Audio PS-1 has the most open architecture of the five processors. The controls for each effect are very sensitive, so it may take you a little longer to get the sound that you're after. But it's worth the trouble. The EQ section is top-notch, the mic preamp sounds good, and the compressor is quite powerful once you get a handle on it. The front and back panels are easy to navigate, and I appreciate having the flexibility of using the dynamics processor, EQ, and mic preamp independently. And you can add an A/D converter to the PS-1 for direct-to-digital recording.

Drawmer's MX60 straddles the line between ease of use and tweakability. The compressor is very smooth and sets up easily; the gate is a little touchy, but it's musical once it is locked in. My favorite feature on the MX60, however, is the 3-band Tubesound, which makes this processor stand out from the pack.

The Focusrite Platinum VoiceMaster has the clearest-sounding mic preamp (it's Class A) and one of the smoothest compressors of the bunch. Each effect dials in quickly and performs well. The combination of a transparent preamp and a saturation stage gives the VoiceMaster a wide tonal palette that works well on just about anything you want to run through it.

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Gino Robair *is an associate editor at* **EM**. *Special thanks to Myles Boisen, Jill Garellick, Jeff Casey, Steve O., Brian Knave, and Laura Forlin, and to Elliot Garellick for his patience.*

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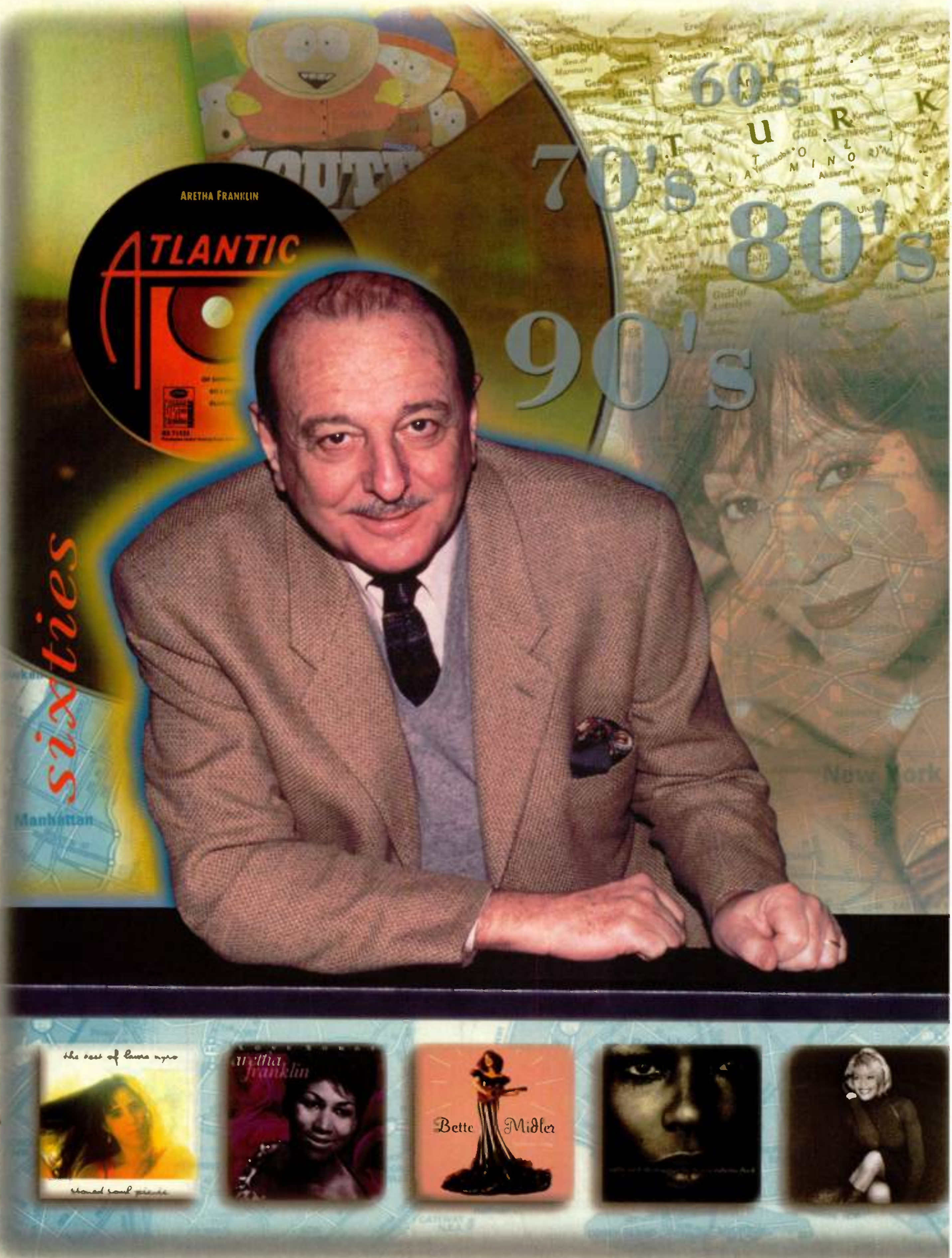


Illustration by Laura Williams

PRODUCTION VALUES

Diva's Choice

Arif Mardin is, quite simply, one of the foremost names in modern music production. Born in Turkey, he emigrated to the United States to attend Berklee College of Music and subsequently joined Atlantic Records (itself formed by Turkish brothers Ahmet and Nesuhi Ertegun) at a time when it was one of the premier labels for jazz. Mardin worked his way up to producing just in time to help Atlantic make a big break into the pop and R&B markets. The list of artists that he has worked with is long and star-studded; it includes the Young Rascals, Aretha Franklin, Bette Midler, Barbra Streisand, Laura Nyro, Willie Nelson, the Bee Gees, and George Benson. Mardin's most recent projects include an upcoming holiday album with Jewel and a song for the soundtrack to *South Park: Bigger, Longer, and Uncut.* But though the range of projects produced or arranged by Arif Mardin is quite broad, he has gained his greatest popularity among singers.

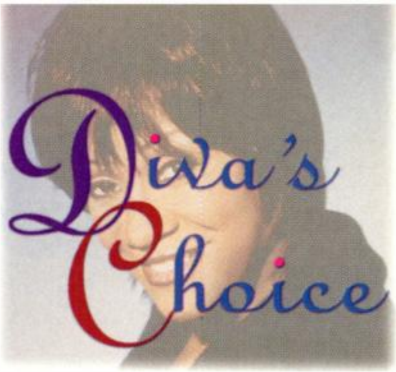
Although he is now most recognized for his production work, Mardin first won notice for his arranging skills, which he has continued to exercise throughout his career. Like Sir George Martin, whom I interviewed for the February 1999 issue of **EM**, Mardin has worked in the studio since mono records were the standard. He has been part of three and a half decades that have brought us to today's 5.1 all-digital world of recording. And Mardin, now a senior vice president at Atlantic, shows no sign of slowing down. His Turkish roots are still a bit audible in his speech after all these years, but he is unquestionably a New York kind of guy.

You came to America to attend Berklee College of Music. How did that come about?

I was a self-taught arranger in Istanbul—asking musicians how high the trumpet can go, that kind of thing. I got a scholarship to Berklee College of Music on the strength of a recording that Quincy Jones had made of three of my arrangement/compositions with a stellar New York jazz band that included Hank Jones and Lee Konitz. Quincy sent the tape to Berklee College, and I got my scholarship because of it.

By Larry the O

Arif Mardin's range of credits is as wide as the Atlantic.



How did you get your arrangements to Quincy Jones?
I met him in Turkey. He came with the Dizzy Gillespie orchestra; he used to write arrangements for Dizzy's big band then. I was arranging for local bands, so when Dizzy's band came, I submitted an arrangement. And Dizzy was kind enough to play it at a rehearsal, just to teach me what a big-band rehearsal was like. I was out of my mind; it was such an honor. I was a big-band groupie; I traveled with the orchestra from town to town and became friends with Quincy. My scholarship to Berklee was called the Quincy Jones Scholarship.

Did you know the Erteguns in Turkey?
No, but they were a very well-known family. The Erteguns' father was a respected ambassador; Washington, D.C., was his last post. Later, I met Nesuhi Ertegun, Ahmet's older brother, who was in charge of the jazz department at Atlantic Records, and he became aware of my arranging capabilities. I was at Berklee at the time and had been writing arrangements for local bands. Two of my arrangements were played at the Newport Jazz Festival. Nesuhi heard them and was instrumental in my getting a three-week BMI scholarship for a wonderful music seminar with Max Roach and John Lewis, the Modern Jazz Quartet, Lee Konitz—all these jazz greats teaching us music for three weeks up in the Berkshires. When the time was right, Nesuhi brought me into the company.

When did you join Atlantic Records?
I came into the company in 1963. It was a lowly job: tape research and unearthing unreleased material by famous artists like John Coltrane and Charlie Mingus. Then I got into more of the administration, and I became studio manager very shortly after that. That's how I learned about quality control, pressings, and recording schedules.

My arranging capabilities were kind of pushed aside because I was doing a lot of production there. Finally, I got gigs writing strings for Atlantic's pop stars, horn arrangements for Wilson Pickett, and things like that. I was an arranger for jazz big bands, but when I started to write arrangements for pop and R&B acts, I couldn't use all those jazz chords; they were strictly taboo for those styles of music.

Then my big chance came when Atlantic signed the Young Rascals. Up until that point, I had been producing and arranging with artists like Herbie Mann and King Curtis, and supervising jazz sessions, but the Rascals' music was the first big-time pop material I had worked on. Veteran engineer Tom Dowd and I were assigned to them as coproducers, and our first record, "Good Lovin'," went number one. That was it. I was bitten by the bug.

How did you become attracted to producing records?
I used to watch how Ahmet Ertegun, Nesuhi Ertegun, and Jerry Wexler produced. I think my first experience of being in a studio during a recording session was in 1963, when Betty Carter was recording *'Round Midnight* with Nesuhi. I thought, "Wow, what a great idea! You're in charge: you're in the control booth, and you can press the button and tell the musicians, "Can you play a little slower?" It was really an eye-opener for me.

The early stuff was all recorded live, wasn't it?
At Atlantic, even in 1963, Tom Dowd had an 8-track machine, and even though he recorded live to a mono tape—that's what everybody did—he also had a backup of the orchestra and the rhythm section separately, and the vocalist on a different track. I wasn't there, but I heard that Ray Charles wasn't happy with the background vocals on "I Believe to My Soul," so he sang all the parts in a girl's range. That was one of the few instances in which multichannel was put to good use in '61 or '62.

So by the time you came to Atlantic, they were already multitracking?
Right, but it was always for a backup or for future stereo mixes when stereo started to become more mainstream. It was never the industry technology of taking a line from one of the takes and flying it into another. Sometimes, though, I saw Lieber and Stoller edit their masters. For example, they would have three takes with the orchestra and the singer, and I remember watching them splice together verses and choruses from different takes.

When you started producing a lot, was stereo already common?
No. We used to mix to mono all the time. For instance, Tom Dowd would often mix the vocalist in live right off the booth, and that would be the master. For the Rascals, though, we put the 8-track to good use: we would record the track and add the harmonies later.

Arif Mardin out on the town with his wife, Latife Mardin, and Whitney Houston.

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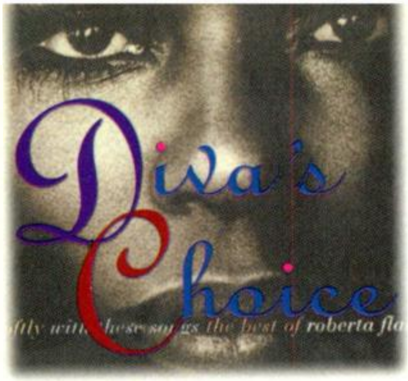


So you got more into multitracking when you started working more with pop than with jazz?

Yes. And then we'd have to go back to our old 8-tracks to make stereo mixes of earlier hits for the Atlantic back catalog.

Can you contrast the way things changed for you as a producer when stereo began to replace mono, versus the way things are changing now as people are starting to mix more in surround? Have you done any surround mixing?

No, but I've been to many demonstrations. First of all, it's great for a record company when you have a new technology, a new product, because you sell more. When CDs replaced LPs, there were a lot of reissues, some of them very badly mastered. But with the early stereo, for example, it was all very gimmicky: you'd have things that would happen on one channel and then move to another channel. I remember there were sound-effects records—trains going by from one speaker to another—and the man of the house would have to try and justify two speakers to his wife: "Look, honey, it's moving." I'm talking about the very early days.

How did you approach stereo when it came in?

Well, it's interesting, because it gave us depth; you could actually move things to the back, and certain things could be in the front. One could make multilayered sound vistas in stereo better than in mono. Definitely, it affected us. I mixed a lot of records for Atlantic at that time. Not only was I producing and arranging, but I was doing my own mixing, too. Some of the mixes that I'm very proud of are Aretha Franklin's *Live at the Fillmore West* and *Amazing Grace*, and King Curtis's *Live at the Fillmore*. These are all live recordings that we made. Instead of taking a clinical approach, we kept a lot of audience and ambience in there. Now, with 5.1, I could do so many things like that. In fact, when the time comes, if they want those records mixed to surround, I would be the first one to try and do it again.

Would you mostly be putting audience and ambience in the surrounds?

Everybody's used to that from the movies. When you go to the movies, you hear whatever sound effects are in the surrounds coming from behind. I might do that with a recording of a live concert. I would make it from the perspective of somebody sitting in the audience and make that the focal point.

However—and I'm just thinking out loud now—if it's a classical concert, and I want to hear certain instruments coming from the back, I would do that too.

Mardin in the studio with Aretha Franklin. The two have worked together on such albums as *Live at the Fillmore West* and *Amazing Grace*.

I don't think there's a law saying that your back speakers must be the audience and applause and sound effects.

Let's take chamber music for an example. A string quartet. It could be gimmicky, but I wouldn't be averse to trying each instrument on a different speaker with, of course, different ambience. Maybe I want to make a circular mix. I could give each instrument its own right, center and left, by spreading the signal, maybe making the signal center in one speaker and then spreading the signal to the next speaker a little bit so you have a bigger sound of each instrument. We're talking about four instruments, so when the violin comes in from left front, it also comes in a little bit from the right front and a little bit from the back speakers. Of course, if it sounds lousy, then I won't do it.

Also, sometimes when they really go into movie mixes, you have dedicated center and dedicated left and right. Why can't we bleed a little bit around? Depending on the songs, sure. With Bette Midler's *Bathhouse Betty,* I did four songs and all of them were distinctly different. One was a very beautiful sort of Hawaiian song called "Ukulele Lady," a kind of novelty, but, at the same time, nostalgic. What can I do with that? I'm not sure. I have to think of having the background maybe bleed to the back speakers so the singers sound interesting. Do I want to put a surf sound in the back? Perhaps I would. For a dance track like "I'm Beautiful," I think the mix should be more precise. It should definitely be more of a true stereo mix; maybe I would not use the back speakers, to keep the mix from becoming too fuzzy.

It's relatively rare in this industry that you get a change as big as surround. And, as with the advent of stereo, for the first few years nobody really knows what they're doing. They're all just experimenting and trying things.

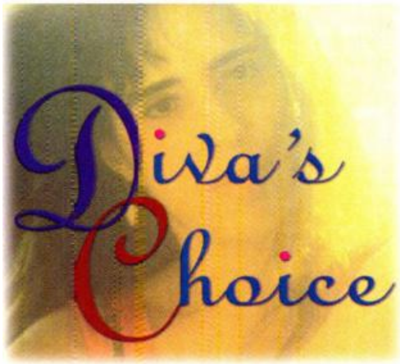
Also, don't forget with the first stereo mixes, the engineer would place the bass on one side and drums on the other. That's just the way it was done. Even at Atlantic, the rule from Nesuhi Ertegun would be, "bass left, drums right," things like that. But then the poor mastering engineer would have so much trouble putting all of that in the grooves. Eventually, we ended up with the formula of kick drum, especially, and bass in the center. It made cutting the records easier, which was the consideration. That, and trying to keep the stylus from skipping out of the groove.

With DVD and surround, obviously, there are no considerations like these. It all has to do with musical considerations and art. I would love to hear Pink Floyd come up with a great new mix of *Dark Side of the Moon.*

I think they are doing a surround mix of that.

And it will be fantastic. I will be the first to go and buy it and tell my wife I need three more speakers.

You have done a huge amount of work producing vocalists. That must be something you really enjoy.



First of all, I am blessed to have worked with people like Aretha Franklin and Roberta Flack. For me, 1997 was like the Year of the Diva: Patti LaBelle, Barbra Streisand, and Bette Midler all in one year. This year, I worked with Brandy and Whitney Houston on the made-for-TV movie *Cinderella*, and I also worked with Diana Ross.

Vocalists like these are so great. There really isn't anything to tell someone like Aretha or Patti LaBelle or Barbra Streisand. At the most, you might give a general idea of what you think, if you're presumptuous enough to speak, and say, "Maybe you could do something like this here, but you know best, darling." She'll do her take and you'll say, "Aretha, this sounds great. We all love it." But she will say, "No, I have to do it again." For us mortals, that was a great take, but she hears something that's even better.

Vocalists can get so close to their work that they lose perspective.

When we would record a track with Laura Nyro, she would turn to me and Felix Cavaliere, my coproducer, and say, "Okay, boys, now you can leave. I want to do my vocals." She was very shy; she would do her vocals at night with the engineer. You'd come in the next morning and there'd be a great vocal there.

Sometimes artists flagellate themselves. The late Dusty Springfield comes to mind. Because of her death this year, a lot of people have asked me about working with her. She was very hard on herself, she would say, "I don't think I'm good. I'll have to do it again and again." Of course, the end result would be great.

Would something like that ever spiral down and destroy the session?

No, no, no. I would definitely be in charge of that. One time when working with Roberta Flack, the engineer gave her such a jolt of feedback in the earphones that she didn't want to sing anymore that day. She was absolutely right to stop, though. She probably had ringing in her ears for a few hours. But that's the only time that I can remember somebody quitting.

Once, in 1966, we accidentally erased eight bars of Aretha's vocals. We were so ashamed, we said, "Aretha, please, could you give us the bridge again? We did something and..." She said, "I sang it. You put it together." She meant that she sang it in a different take. We had recorded on 8-track, but this is before we even dreamed of comping, or making a composite of vocals from several takes. She gave us an order, saying, "The technology exists. Take my vocals from another take and put it there." We did just that. We transferred that vocal from a different take to another tape and then rerecorded, and flew that vocal in, trying to sync it and match the tempo.

You must have had to do that wild, because time code wasn't around in recording studios then.

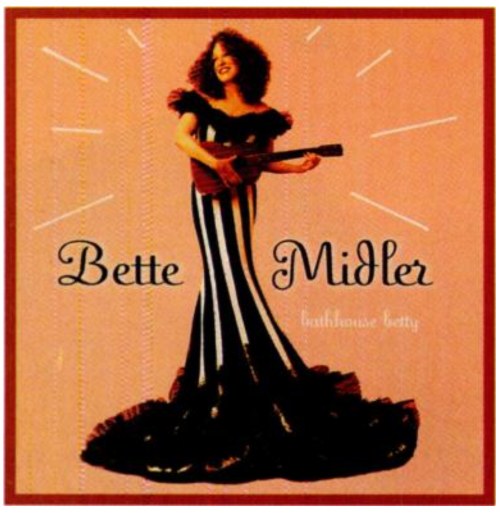
That's right, there was no time code, and of course, the takes were different tempos and we had to fly the vocal in phrase by phrase. Now we do that as a matter of course. I don't have to name names because everybody does it—you grab a good verse from one take, a good line from another.

Mardin produced "Eyes of a Child," written by *South Park's* Trey Parker and sung by Michael McDonald. The track was used for the soundtrack to the movie *South Park: Bigger, Longer, and Uncut*.

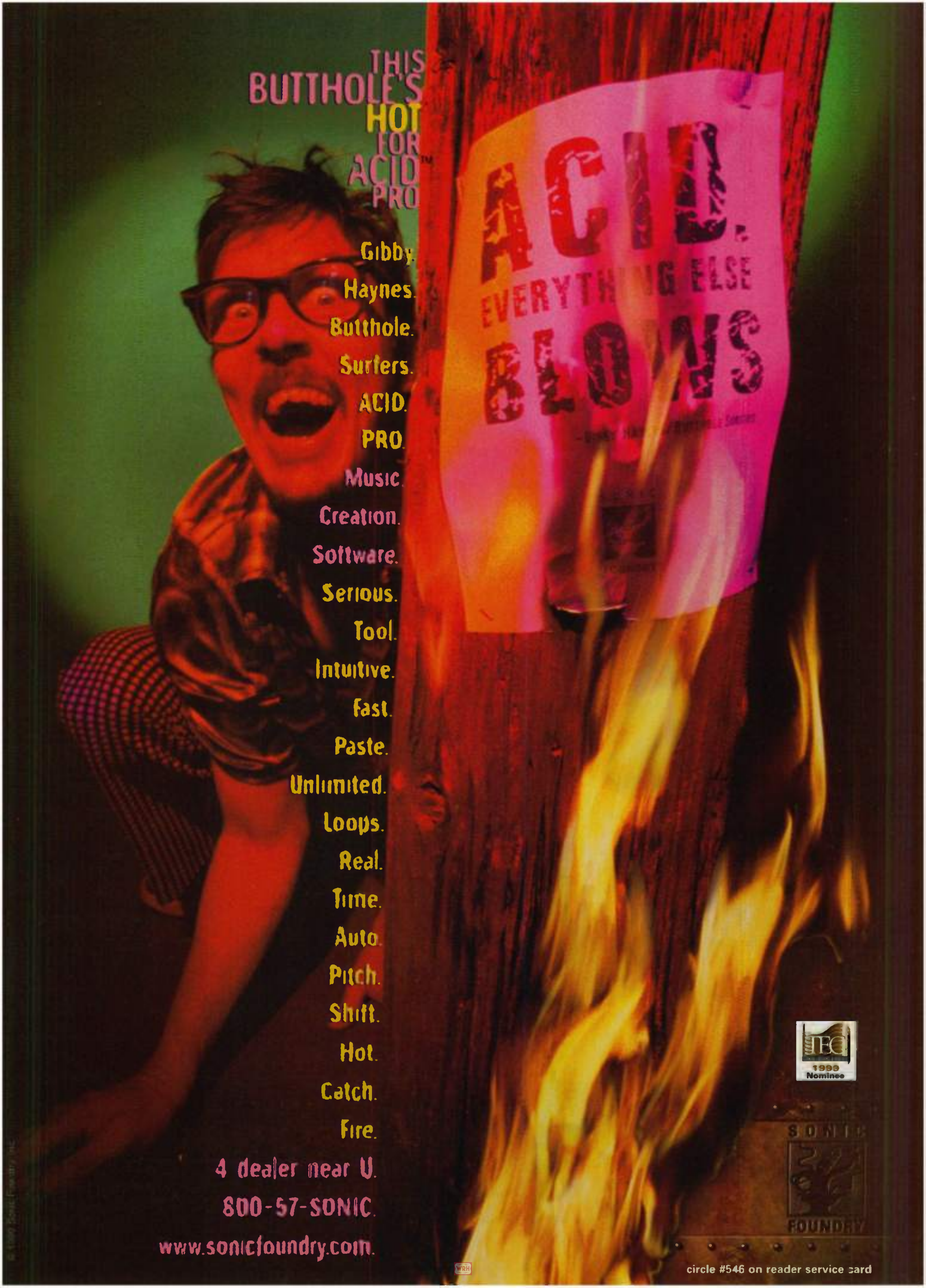
Perhaps you can give me an example of how you have used technology to achieve something, and one of how you eschewed it in favor of something much simpler and more crude.

The best way that I have used technology occurred just about a week ago. For Barbra Streisand's latest album, we recorded the tracks in L.A., and then brought the tapes to New York to mix with Frank Filipetti. She likes to hear the work in progress, so we used the EDnet [a codec that delivers full-bandwidth stereo audio in real time over ISDN lines] to send the mix to her in Malibu; she has a pair of Genelec monitors in her living room. We'll make a date of it: "Okay, at such and such an hour we will call you on the EDnet," and she will be in her living room listening. She can give us mix instructions like, "Okay, I think I want to hear my voice a little louder here." So this is one way of really using technology. We're working on an automated, digital Neve Capricorn board, too, so it's no big deal; the assistant engineer doesn't have to go back to the notes and spend three hours recalling a mix. Touch a button, and it's there.

Sometimes I don't care to stay in the digital domain all the time, and there are times when Pro Tools and computers would actually take longer. I have an analog "idiot device" called a Vocal Splicer. It was a late-'80s thing that David Foster and a lot of L.A. producers used to use. It's like a little six-hundred-dollar crossfader box. I use



For Bette Midler's *Bathhouse Betty*, Mardin produced four songs, including the whimsical "Ukulele Lady."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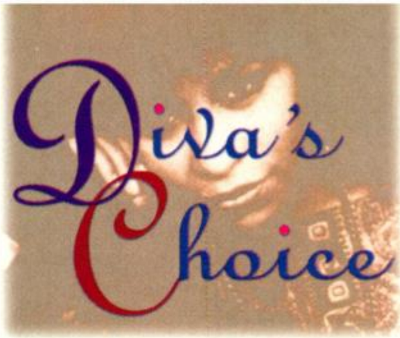

that to splice together vocal parts, and sometimes I can do in one hour what would take state-of-the-art technology three hours. I'm like the mark of Zorro—zoom zoom zoom on the box and it's finished. But, of course, the other way is more accurate, more scientific.

When you're recording vocals, especially now that things are all multitracked, at what point do you put the lead vocal down? Do you usually have all the rest of the tracks done, or do you record the lead vocal over rough tracks and then replace them?

I must have a rough vocal by the artist to give me a simulation of where the dynamics will happen. If it's a synthesizer track, I may even put down the rough vocal before we go to the programmer's home studio. Sometimes, if the artist is not available, I get a session singer and bring her into the programmer's home studio and I'll get somebody else singing. At least I have something. I hate to do work in a void.

You put the rough vocal over simple bass and drums and a couple chords?

No, no, no. At least a 70 percent finished arrangement.

If we're recording the instruments live, it's good to have the vocalist there, obviously. For example, on the *Bathhouse Betty* jazz ballad "Sold My Heart to the Junkman," Bette was there singing with the rhythm section, and when we selected the take [to keep], we used most of that vocal. You have all the rubato feeling.

I see primarily music albums on your discography, but there are also many other things like the *Cinderella* television special, and the soundtrack recordings for *Rent* and for *Why Do Fools Fall in Love*, the film about the life of Frankie Lymon. How does your job differ when you're producing that kind of session?

For something like *Rent*, logistics is a big issue—trying to get the best performance out of the singers in a short time—because Broadway cast albums you have to do very quickly. So you have a certain order: start with the largest ensemble, then go to the solo pieces. You have to be a taskmaster because things must be done efficiently, yet in a very musical way. Both musicals I did almost like a pop recording: part of each was done live, the other part in layers, where you do the track first and then bring the vocalist in.

Usually, these musicals are recorded on the off nights [from theater performances]. The musicians come in and record the music from beginning to end, maybe twice, and then the producers and mixers pick out of the choices. I did mine a little differently. We recorded the tracks on some songs and put the vocals on later; other songs had to be done live for tempo changes, and then we had enough iso booths to be able to redo vocals, if needed. I'm kind of proud of *Rent* because the sound is alive, but I was also able to get the best vocals.

So what do you think of the new "swing revival" (which seems to me more like jump blues than swing)?

I really love it. Atlantic Records signed a wonderful group called Atomic Fireball, and I did a song with them. They were so fabulous because these are young guys, but they're not faddists. They love their grandparents' music; that's what it's about. They listen to old records—to Louis Jordan and Cab Calloway—and it shows in their music. You know what? Sometimes I don't want a whining baritone talking about gloom and doom. These people want to dance and have a good time, so I think it's a great thing.

Even with all your producing, you still keep up your arranging, too.

In 1993 I had a flamenco jazz piece commissioned. And this year I wrote a Duke Ellington medley, three of Ellington's compositions: "Koko," "Things Ain't What They Used to Be," and "Carnegie Blues," which was part of *Black, Brown, and Beige*. It's a fabulous composition; Ellington was experimenting with the sound of those dominant seventh, raised nine chords around when he wrote *Black, Brown, and Beige*.

I also wrote a big-band arrangement for what turned out to be Eddie Harris's last gig. So I keep on charging my batteries. I also write for big orchestras: I did that for Barbra Streisand on both albums. I wrote for a 60-piece orchestra, which is a great challenge. I wrote something for Carly Simon in the film noir style, really 1940s.

In the meantime, of course, I also arrange for my pop records. The only thing is, it's not like the old days anymore, where you wrote an arrangement and the rhythm section played it. You cooperate with a programmer, if it's a synth kind of thing. I'm there while we program and I give my input, working and arranging as we go.

You've accumulated such a wealth of production knowledge. I wonder if you have any comments for EM readers.

Mardin with fellow producers Frank Filipetti (left) and Phil Ramone (center).

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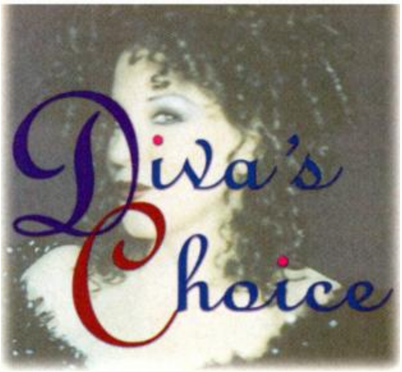


When it comes to producing music, the song is the most important thing. The vehicle is very important. You have to be conscious of lyrics. Sometimes they're nonsensical, sometimes they're very important.

When I say nonsensical, it could be like, "Hey, baby, let's dance." That has it's place: sometimes you like a hamburger, sometimes you want filet mignon. I'm not putting down anything. The only thing is, you have to be aware of the lyrics, of the artist's capabilities, the artist's ability to project to a segment of the record-buying public. If you're working with an established artist, you need to know who's going to buy this record. Of course, I'm really being mercenary now. There are a lot of considerations, but possibly the most important thing is to get 200 percent from the artist. Get a lot.

Another important thing is to select the right key for the song. Sometimes, a guitar player will play in E major because it's great on the guitar, but the vocals may sound much better when the song is in E-flat.

Is there anything else you'd like to add?
One important thing about vocalists: the reason that I get along with them is because I respect their genius. I never challenge a person like Bette Midler or Aretha Franklin. If they say, "Can we try this here in the production?" I wouldn't say, "What would she know?" On the contrary, thinking of the body of work these people have done, she may well know something. Okay, let's try it. It didn't work? Fine. Either it works or it doesn't work. No ego, nothing. Respect. I think it's because they realize that I respect them and their art that we get along fine. That's the key, I think, for every producer: don't look down. Respect who you're working with.

Larry the O *is a longtime contributor to both* **EM** *and* Mix *magazines, as well as a musician, sound designer, producer, engineer, and maker of magnificent mochas.*

Talk Is Cheap

You can teach an old computer new tricks.

By David Rubin

As any *Star Trek* fan will tell you, computers are supposed to listen when you speak to them. Issue a command ("Computer, run a level 1 diagnostic"), and the attentive circuitry responds appropriately. In one memorable scene from a *Star Trek* movie, the crew had traveled back in time, and Scotty, seeing a Macintosh for the first time, picked up the mouse and spoke into it, thinking that it was a microphone. Audiences laughed knowingly, but that telling scene unearthed a deeper question: will the computer mouse eventually be replaced by a microphone as an input device?

JACK DESROCHER

My guess is that it won't happen right away. After all, we're still typing words and commands on a keyboard that was modeled after a product of the industrial revolution. Nevertheless, voice-recognition technology has made great strides in recent years and is already in widespread use on both the Mac and PC platforms.

In fact, with a little ingenuity, you can operate your desktop studio by merely barking commands into a microphone. Imagine standing several feet away from your computer with both hands on your MIDI guitar controller and recording, playing, stopping, and rewinding your favorite sequencer simply by speaking into a headset mic. Perhaps you'd rather use your voice to call out note durations while step entering music in your notation program. Voice-recognition technology holds great potential for musicians, and it's relatively easy to get started using. I'll explore some general issues and then provide step-by-step instructions for using the Mac's PlainTalk software as an example of how speech-recognition technology can be applied in the studio.

Before we continue, however, let's distinguish between the two main categories of speech-recognition systems: dictation programs and voice-command programs. Dictation programs

convert spoken words directly into text in nearly real time. They can be a great aid to people with disabilities, and they're also gaining popularity in the business community and in professions where it's hard to work and type at the same time. Several companies now offer sophisticated dictation programs for the Windows platform, including Dragon Systems (www.dragonsys.com), which sells a range of programs at different prices and for different purposes; and IBM (www.software.ibm.com/speech), which markets a line of products based on the company's own ViaVoice technology.

Macintosh users have fewer options when it comes to dictation software, but the Mac does have one big advantage over Windows PCs: its operating system software includes a sophisticated and surprisingly robust speech-recognition technology called PlainTalk. PlainTalk doesn't perform dictation, but it does offer voice-command ability. This means that it can recognize spoken words or phrases that then trigger actions or initiate commands. (Actually, PlainTalk itself doesn't initiate commands; it simply tells a client application what it heard, and the app decides what to do about it.) PlainTalk is just what you need to operate a desktop studio, and it's readily available to anyone with a Power Mac and Mac OS 7.5 or later.

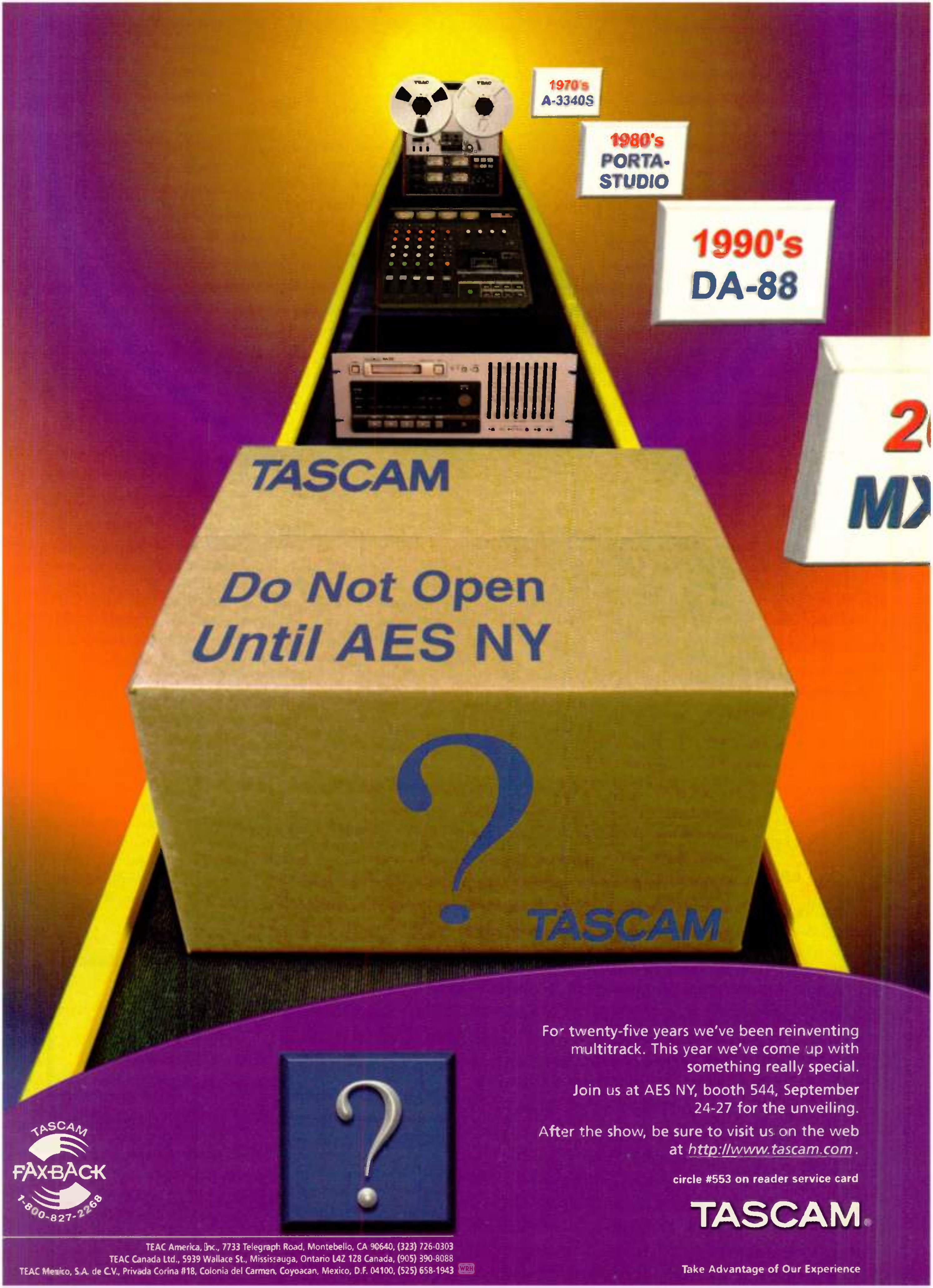
Let's take a closer look at the Mac's approach to speech technology and how to harness it for musical purposes. Windows users should contact the companies mentioned earlier to see what resources are currently available for voice-command operations on that platform.

PLAIN TALK ABOUT PLAINTALK

Apple's PlainTalk software actually covers two areas of voice technology: speech recognition and speech synthesis. Speech synthesis is essentially the opposite of dictation; it converts text into spoken words (synthesized speech). Mac users can choose from several voices that are included with the operating system software and can have text read to them from SimpleText documents or dialog boxes. Speech synthesis, however, is of limited value in the studio. In fact, I recommend disabling PlainTalk's text-to-speech option so that you won't have annoying alert dialogs read to you when you're trying to concentrate during a recording session.

PlainTalk's speech-recognition component, on the other hand, can enhance the Mac's graphic interface by freeing your hands from the tyranny of the keyboard and the mouse. Moreover, PlainTalk boasts several features that add to its effectiveness and make it easy to use. For example, it is a speaker-independent system. Unlike some applications that must be "trained" to recognize your voice, PlainTalk works with any male or female adult speaker of North American English. Voice samples of more than 500 adults from different parts of the continent were used to develop the acoustic models, which the software uses to recognize speech patterns. (According to Apple, PlainTalk may not work reliably if you speak with a heavy accent, and it doesn't work well for children because their speech is composed of different spectral characteristics than adults'.)

PlainTalk is a "continuous-speech recognition" technology, which means



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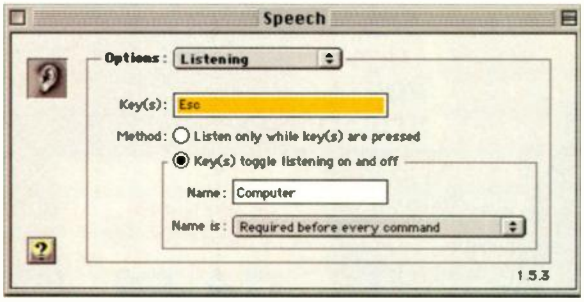

FIG. 1: The Speech control panel enables you to turn Speakable Items on and off and to specify a trigger key or word that tells the computer to listen for commands.

that you don't have to add ... a ... pause ... between ... words when you speak, unlike earlier "isolated-word" systems. In fact, PlainTalk works best if you speak clearly and naturally in fluent phrases or sentences. PlainTalk is also tolerant of extraneous noises (such as coughs or door slams) and of different kinds of acoustic environments (a studio versus a conference room, for example). In addition, the program uses a flexible "finite-state" grammar, which defines the commands it will respond to. You provide PlainTalk with a set of phrases (or sentences or word groups) to listen for; other speech patterns are simply ignored.

Actually, you don't have direct access to PlainTalk itself; it's simply a set of programming tools that control speech recognition. To use it, you need an application that incorporates PlainTalk speech-recognition code. Fortunately, Apple includes a simple Finder utility called Speakable Items, which you can start using right away.

SPEAK EASY

Speakable Items works by scanning the contents of the Speakable Items folder (located inside the Apple Menu Items folder). When it hears the name of one of the items in the folder, it responds by double-clicking that item. You could, for example, create an alias of your e-mail program, rename it "get my mail," and drop it into the Speakable Items folder. Then whenever you say, "Get my mail," the Speakable Items utility launches your e-mail program. It's that simple.

I was recently working on an audio file in BIAS *Peak*, and when I finished my editing for the day, I saved the audio file as a *Peak* document. Next, I made an alias of the file, named it "open my demo song," and placed the alias in the Speakable Items folder. Now when I boot my computer in the morning, I simply say, "Open my demo song," and the computer automatically launches *Peak* and opens the audio file in the waveform display. What's more, I don't have to dig through multiple folders to locate the file or the application.

Of course, to use Speakable Items, it must be installed on your Macintosh. If you're not sure that it is, check the Apple menu and see if the Speakable Items folder is listed. Also, check the Extensions folder to verify that the Speakable Items, Speech Manager, and Speech Recognition extensions are present. Next, check the Control Panels folder and locate the Speech control panel (see **Fig. 1**). From the drop-down menu, choose Speakable Items, and then click the button that turns it on. You must also plug a microphone into the Mac (unless your monitor has one built in) and set the input device to External Mic in the Monitors & Sound control panel.

If Speakable Items is not already on your computer, you can install it directly from the Mac OS 8.5 CD-ROM. Open the English Speech Recognition folder, located inside the disc's Software Installers folder, and double-click the installer icon. If you don't have the Mac OS 8.5 disc, you can download the necessary software from Apple's Web site (www.apple.com/macos/speech). The version of PlainTalk that comes with Mac OS 8.5 (version 1.5.3) offers better performance than the earlier versions; it does not, however, support the iMac or the newest G3 computers. An update is currently available with Mac OS 8.6 (free to Mac OS 8.5 owners at Apple's Web site).

So you'll know that the computer is listening to you, the Speech Recognition extension includes a floating Feedback window (see **Fig. 2**). This window displays an animated character (there are several to choose from—my favorite is Phil) and a text field that shows the commands to which the computer has responded.

The Speech control panel lets you determine how and when the computer listens to commands. One option, which Apple calls the "push-to-talk" method, lets you assign a key that triggers the listening mode; the Escape key is the default setting. With this option, the computer listens for commands only when the assigned key is held down. This approach affords maximum control over the voice-recognition activity and minimizes "misfires" caused by incidental conversation. But it ties up one hand and forces you to stay close to the keyboard, so it's not always the best choice for a desktop studio.

The option that I prefer uses a spoken word (or phrase) instead of a key. With this method, the computer listens all the time but responds only to commands that follow the assigned word. "Computer" is the default word, and it works quite well for me. So when I open the audio file mentioned earlier, I actually say, "Computer, open my demo song." Having a special trigger word reduces misfires, although it's still not as effective as the push-to-talk method. You'll get the best results if you choose a word or phrase that isn't likely to come up during normal conversation. Also, avoid words that sound like other common words that you would frequently use.

The least reliable option does away with trigger words and keys altogether. The computer simply listens all the time and tries to respond whenever possible. If you're working alone, this method might work fine, but it's generally more prone to misinterpretation. And if there are other people (that is, other voices) in your studio, it can definitely lead to problems.

FIG. 2: The Feedback window includes an animated character (this one is Phil) who responds to your commands. A text field shows the recognized commands that are received. The word "Computer" beneath Phil shows the currently specified trigger word; the sound waves emanating from the left indicate that the computer is currently receiving audio input.

PC or G3

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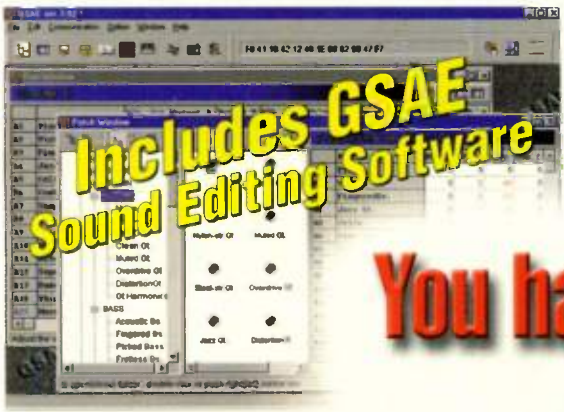

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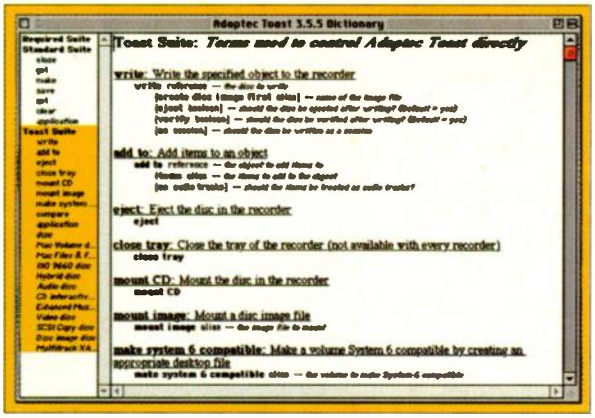


FIG. 3: AppleScript-enabled programs always include a dictionary showing commands and their associated scripts. This is the AppleScript dictionary for Adaptec's *Toast*.

SAY WHAT?

Once you have Speakable Items up and running, you can explore its potential as a control interface for desktop studio operations. As I described earlier, you can open documents and folders and launch applications with Speakable Items, but that's only scratching the surface of what this utility can do. That's because Speakable Items can also be used to launch AppleScript scripts.

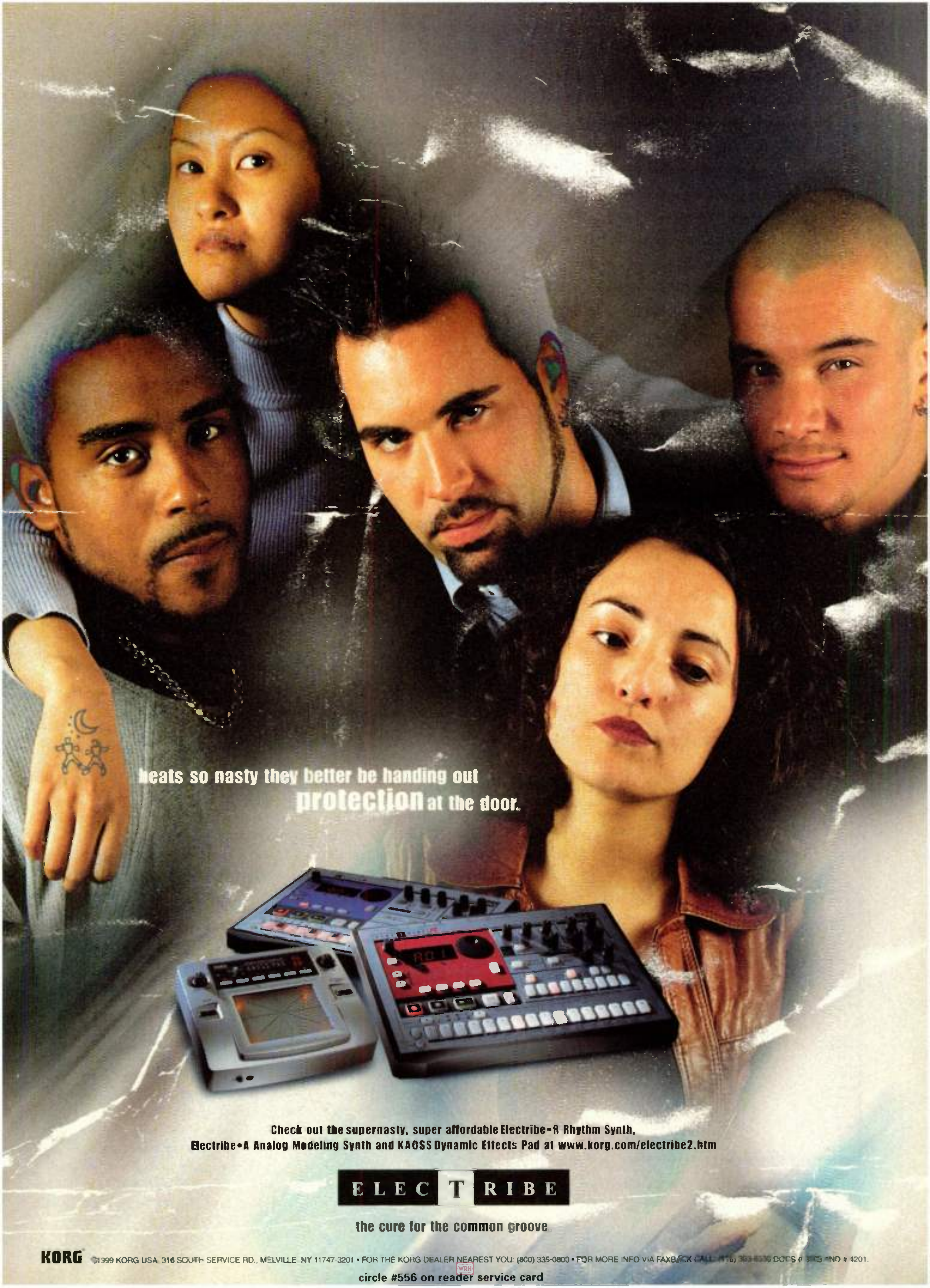
AppleScript is Apple's macro-creation and automation language. It enables you to perform multiple operations by simply double-clicking a script icon or accessing a script from within an application. The Speakable Items folder includes several scripts that perform such simple tasks as closing all windows, setting the computer's volume to maximum, or viewing a window's files by date. There's even a script to make a new file into a "speakable" item. (Just select the document and say, "Computer, make this speakable," and you're done.)

Scripts are created and edited in the Script Editor, which resides in the AppleScript folder inside the Apple Extras folder. By combining AppleScript with Speakable Items, you can gain voice control over a wide range of functions in any program that supports AppleScript. You can tell if an application is "scriptable" by dropping the application icon onto the Script Editor icon. If the program is scriptable, a window will open showing the program's AppleScript dictionary (see **Fig. 3**).

With a scriptable program, you can create a macro by entering text directly into the Script Editor. You might, for example, create a script that tells your sequencer to start recording. You would then save the script and name it "start recording." If your trigger word is "computer," you would simply say, "Computer, start recording," and your sequencer would enter recording mode. There are several good books that give a full explanation of how to use AppleScript and the Script Editor. You can also get detailed information about using AppleScript by visiting Apple's Web site (www.apple.com/applescript).

NOT MY TYPE

Most current audio and music programs have no direct support for AppleScript. There is, however, a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work-around for this problem: you can use a PlainTalk-enabled macro program to fake a keyboard command in response to a spoken word or phrase. Some macro-creation programs, such as *QuicKeys*, are AppleScript compatible, so you can use them with Speakable Items to trigger keyboard events that affect the frontmost application—your sequencer, for example. For an even easier (and cheaper) solution, download a copy of Michael Kamprath's *Speech Typer* (www.kamprath.net/claireware/speech_typer.html), a simple $15 shareware utility that converts spoken words into keystrokes.

Speech Typer consists of two parts: the Engine, a background-only application that does most of the work; and the Controller, which is used to edit phrases and set preferences (see **Fig. 4**). The *Speech Typer* Engine listens continuously. When it recognizes a phrase from its Listen Phrase list, it types the corresponding user-defined Response Phrase, which can be anything from a single keystroke to a full business memo. (*Speech Typer* uses the same trigger word that you have set up in the Mac's Speech control panel.) In many ways, *Speech Typer* is similar to Speakable Items, except that you're sending keystrokes to the computer rather than double-clicking items. And that's just what you need in order to operate your favorite sequencer with voice commands.

Most high-end sequencers (such as MOTU *Digital Performer*, Steinberg *Cubase VST*, Emagic *Logic Audio*, and Opcode *Vision DSP*) allow you to assign keyboard commands to trigger almost any function in the program (see **Fig. 5**). You can cut, copy, and paste; operate the transport controls; and select note values for step entry and quantizing. Unfortunately, *Speech Typer* doesn't use the Mac's Control, Option, or Command keys, so you'll probably have to reassign many of the key commands in your sequencer. Nevertheless, there are plenty of keys to go around for most tasks. For example, I set up *Cubase VST* with the following key equivalents: Record = R, Play = P, Stop = S, Rewind = W. Then I set up *Speech Typer* to type those letters when I speak the corresponding words. Now I can sit with both hands on my MIDI guitar controller, and when I say, "Computer, record," *Cubase* goes into Record mode and I can start playing. I can also perform other common tasks, like turning the metronome on and off, by expanding my list of commands.

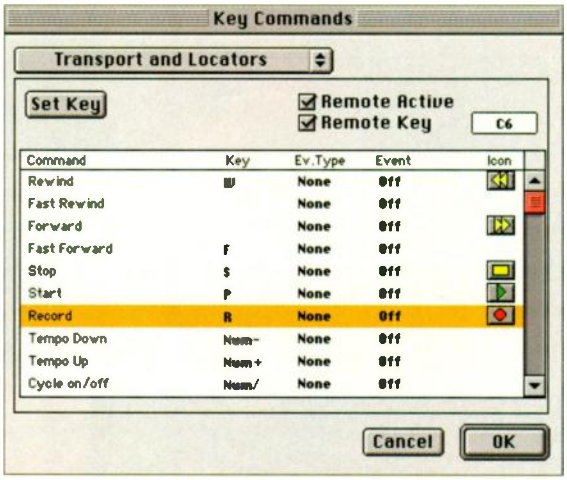

FIG. 5: Most sequencers (*Cubase VST* is shown here) provide a window where you can assign keystrokes to commands and transport functions. By combining this window with *Speech Typer*, you can "speak" your keystrokes instead of typing them.

STATE OF THE ART

Speech technology does work, and it's lots of fun to play around with, but it's still far from perfect. Furthermore, it may be impractical in many studio settings. For example, if you record directly into an audio-editing program or audio/MIDI sequencer, you can't tie up your computer's Mic input for issuing voice commands. In fact, some music and audio programs are simply incompatible with PlainTalk.

You'll also have to get used to a bit of lag time when controlling a sequencer by voice. When you speak a command, *Speech Typer* must listen and recognize the speech pattern by comparing it to its list of commands. It then types the keystroke, which the sequencer must recognize and respond to. Each of these steps takes time; you can reasonably expect a one- or two-second delay after you speak a command before things start to happen. That pretty much rules out things like punch-in/out recording, where speed and precision are essential.

Moreover, *Speech Typer* doesn't always recognize my commands the first time around, so I often have to repeat myself. I find that enunciating clearly improves my success rate, although it's never 100 percent effective. Some words consistently give me trouble and have to be changed to more easily recognized phrases. For example, "rewind" seems to cause confusion, so I use the command "go back," which works much better. In addition, some microphones work better than others. I found that Apple's PlainTalk mic worked the best of several inexpensive dynamic mics that I tried.

In spite of their shortcomings, however, Speakable Items, *Speech Typer*, and other PlainTalk-enabled programs hold much promise for the future. Apple has made a serious commitment to speech technology (it has been shipping speech-related products since 1993), and you can expect to see more widespread use of voice-recognition tools in the coming years. For now, you can enjoy a glimpse of what lies ahead in computer interface design. And just remember: the next time you start to curse at your computer, it might be listening.

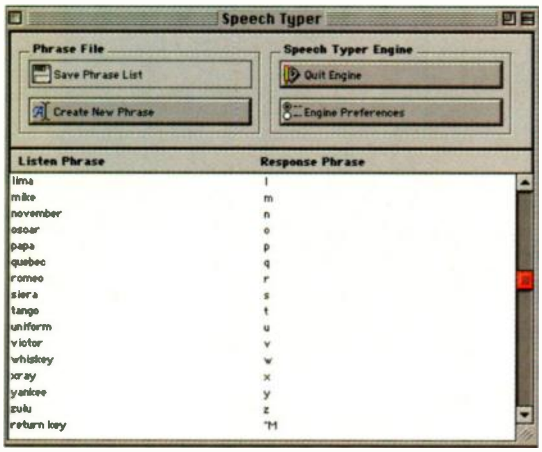

FIG. 4: The easy-to-use shareware utility *Speech Typer* enables you to convert spoken words and phrases into one or more keystrokes.

Associate Editor **David Rubin** *talks to his computer only when he has something important to say. Special thanks to Tom Bonura of Apple Computer for his help in preparing this article.*

Picture Perfect Sound

Keeping film, video, and audio in harmony.

By Gene Takahashi

As film, video, and audio converge in computer-based workstations, individual users are forced to wear a steadily increasing number of hats. Traditional barriers that once clearly separated jobs in sound, music, and picture are starting to crumble, and musicians have a growing need for knowledge in related fields.

Whether you produce sound effects, music, or dialog, as an audio specialist you need to understand how audio works in relation to film and video so you can properly prepare your materials. Furthermore, a well-rounded knowledge of technical issues and terminology will make you a better team player on audio-visual projects. Let's take a close look at how film and video are related, and how that relationship may affect synchronized audio tracks.

PETER HORVATH

FILM VS. VIDEO

As most people know, film is developed and projected in much the same way as the slides you take with your personal camera, whereas video records images as magnetic signals on tape. Film's picture resolution is quite high, so it typically presents viewers with a richer visual experience than video. Vibrant colors, fine details such as hair, and subtle lighting are much more apparent and well defined on film. Because of this better picture quality and film's "analog" look, most productions that have the budget for it use film.

Video, on the other hand, is quicker, cheaper, and easier to work with. To save time and money, most film productions are initially edited on video. Film is shot, developed, and then transferred to video workstations, where editors and directors can make choices about which shots to use. Once they've decided on most of the changes, the workstations generate a "cut list," and the film is spliced together by hand,

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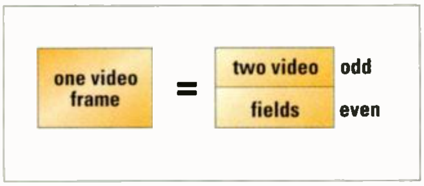


FIG. 1: One video frame is made up of two interlaced video fields.

using the age-old tools, razor blades and tape.

Whenever you switch between video and film, you have an instant problem: *frame rate.* The frame rate is the number of individual pictures shown each second (measured in *frames per second,* or *fps*). Video standards around the world use various frame rates from 25 fps (PAL and SECAM) to 60 fps (HDTV). The television standard we currently use in the United States, called NTSC, was adopted by the FCC in 1953. (Of course, the standards are changing as digital TV emerges, but that's another article.)

NTSC video flashes around 30 individual frames every second, whereas with film, you get only 24 frames per second. In addition, video subdivides each frame into two smaller segments, called *odd* and *even fields,* for a total of 60 images per second (see **Fig. 1**). Obviously, playing a 24 fps theatrical feature on a 30 fps VCR requires some adjustment.

PICTURE PULL-DOWN

Film is transferred to video in a *telecine bay,* a very expensive piece of equipment that few of us will ever have the pleasure of owning. "Telecine" (pronounced te-le-SIN-ay) was coined from the words "television" and "cinema." You must understand some of the telecine process in order to understand how audio works with film and video.

A direct frame-for-frame transfer between film at 24 fps and video at 30 fps would be problematic. The videotape would run 6 frames too fast every second, thereby shortening the overall picture by 21,600 frames (12 minutes) every hour. The audio tracks would go out of sync almost immediately on playback.

A clever technique called *2-3 pull-down* solves the problem. Instead of making each frame of film one frame of video, the telecine process takes four frames of film and makes them into five frames of video. The first frame of film is pulled down into a projector, and two video fields are electronically recorded onto tape. The second film frame is then pulled down and recorded as three video fields. The same process is repeated for the third and fourth film frames. Because frames of film are recorded in alternating groups of two or three video fields, we call this process "2-3 pull-down" (see **Fig. 2**).

The ten fields recorded from the four film frames make up five video frames. (Remember this four-to-five ratio; it's important.) In every second, this adds 6 frames to the original 24 frames, producing a 30 fps video copy that maintains the timing of the original film.

AUDIO PULL-DOWN

If you want to edit sound to film, understanding the pull-down process is very important because most sound and music editors edit their sounds to video. (They don't have the luxury of building movie theaters in their homes.) They lay their sounds to 30 fps videotape that has been "pulled down" from the 24 fps film for the big screen.

So what happens when you take the golden soundtrack you synched to video and attempt to play it with film? The sound should be in sync with the picture, right? Unfortunately, that's not what happens. In fact, the audio begins to drift, and the longer you play it, the more out of sync it gets.

But wait a minute. Didn't I just finish explaining how 2-3 pull-down keeps the video copy in sync with the film? Well, I lied, but just a little. I stated that video runs at *around* 30 fps. It actually runs at 29.97 fps because of a workaround that was introduced when color broadcasting was invented. That means film frames don't break down so neatly into video frames. During the telecine transfer process, the playback of film is slowed down a bit to maintain our four-to-five frame ratio. The ratio is actually 23.976 film frames for every 29.97 frames of video—subtracting 0.1 percent (the difference between 30 fps and 29.97 fps) from 24 fps gives us 23.976 fps (see **Fig. 3**).

This means that when you look at film transferred to videotape flashing 29.97 fps on your VCR, you are actually seeing 23.976 of the original film frames per second. If your audio is still traveling at 24 fps, it is moving too fast and will run ahead of your picture. The audio must be slowed down to match the actual rate of your picture. This is where your audio editor's sample-rate pull-down command comes into play.

By activating the pull-down function, you slow the audio playback rate down by 0.1 percent to match the video. On your DAW, your sample-playback rate will change by 0.1 percent (44.056 kHz instead of 44.1 kHz). *Film rate* refers to sound that was recorded in sync with

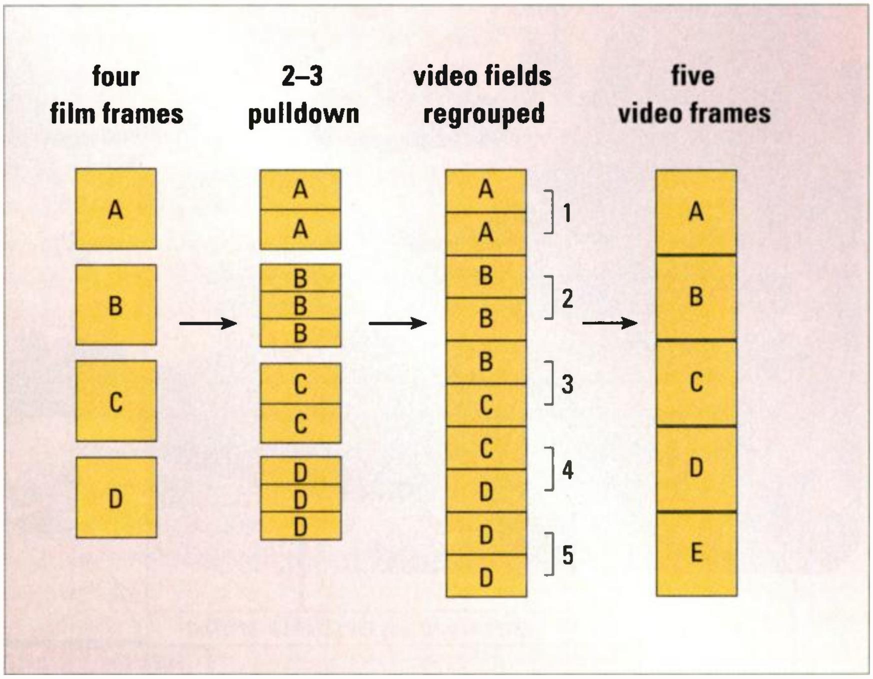


FIG. 2: The SMPTE-A 2-3 pull-down transfer process converts four film frames into five video frames.

$$\frac{4}{5} = \frac{24\text{ fps}}{30\text{ fps}} = \frac{[24\text{ fps} - (24\text{ fps} \times 0.001)]}{[30\text{ fps} - (30\text{ fps} \times 0.001)]} = \frac{23.976\text{ fps}}{29.97\text{ fps}}$$

FIG. 3: This formula shows how the four-to-five ratio is maintained even though the video speed is actually slightly less than 30 fps.

the film camera, while *video rate* is defined as the pulled-down rate of the sound.

TIME CODES

It is important to realize that time codes and frame rates can be, but are not necessarily, the same thing. In general, music production uses 30 fps time code because there is no need to sync to picture. Video production and television broadcast, however, use a form of 29.97 fps time code because all timing is dependent on the video frame rate. Film productions typically employ the same time codes used by music or video productions.

Because of the difference in synchronization schemes used by film cameras and field audio equipment, it's common to have a picture frame rate of 24 fps and an audiotape recorded with 30 fps time code. Some productions may not synchronize their camera and sound equipment. In that case, the production sound mixer simply stripes the tape at (hopefully) 30 fps and lets the audio equipment run free. Time code alone will not tell you whether you're running at film rate or video rate. The actual picture frame-rate information must also be known.

RECAP

When you have field audio (such as dialog) recorded at film rate (24 fps) and you want to edit that audio to pulled-down video (29.97 fps), you must set the pull-down to slow the audio playback (by 0.1 percent) in order to maintain sync with the original film frames (which have been slowed to 23.976 fps). The reverse is also true. When you have sound effects and music edited to telecined video (23.976 film fps), you must speed up the audio (by 0.1 percent) to maintain sync with the projected film (24 fps).

Most professional workstations have appropriate settings for dealing with any pull-down situation, but the operator must know at what rate (video or film) the source tapes were made.

TERMINOLOGY

Unfortunately, pull-down terminology is not standardized, and confusion may arise when you try to decipher various owners' manuals. Manufacturers generally lump the topics of picture-frame manipulation and audio-playback speeds under the general heading "Pull-down." You are expected to know the difference (even if they don't) when they discuss the issues.

I've run across various terms for the pull-down settings, including "pull-in and pull-out," "pull-down and pull-up," and "1.00 and 0.99." Time-coded DATs may force a pull-down when a tape striped with 30 fps time code is inserted into a machine set to 29.97 fps time-code settings. Most users' guides do not give a detailed explanation of the process, but some explain the relationship between the settings and the sample-playback rates. Armed with an understanding of pull-down and with your manual, you should be able to figure out proper settings.

ALL IN HOW YOU LOOK AT IT

When a project involves film or video, timing is derived from the visual elements, and the audio has to be adjusted to achieve synchronization. If you're doing audio work that involves both visual formats, you must know how to deal with the differences.

The ability to go directly from DAW to mixing is now a reality, and this method will grow in popularity as production budgets are squeezed. There never seems to be enough time, and that causes a lot of frayed nerves as a project nears the end of the post-production process. By remembering the differences between audio-visual formats and dealing with them correctly, you can avoid the frustration of unnecessary delays.

Gene Takahashi *works for Pixar Animation Studios. He is also starting up MultiMedia Audio Productions, a provider of audio services for corporate multimedia.*

Hella a Cappella

Capture evocative musical portraits of vocal groups.

By Brian Knave

The more I work with electronic instruments, the more I appreciate acoustic ones—the human voice, in particular. The voice is not only the oldest musical instrument, but the most complex, flexible, and—thanks in part to its connection with language—expressive. What other instrument, whether acoustic or electronic, has such a wide range of timbre and emotion?

By excluding other instruments from the mix, a cappella music highlights the expressiveness of the voice and multiplies it by the number of singers in the group. The results can be greatly satisfying musically—in terms of soul *and* technique—not to mention a welcome change from the technology-driven soundscapes that electronic musicians often inhabit.

I recently recorded Wicki 6, a sextet from Berkeley, California, and I was reminded of the musical richness, range, and potential of a cappella. But I was also reminded of the many critical considerations and technical challenges that await the engineer attempting to turn out good recordings of vocal groups.

A matched pair of Earthworks SR77 cardioid condenser mics captures the vocal stylings of Wicki 6 (from left to right: Elaine Chao, Bhakti Klein, Amanda Weeden, Dubie Dubendorfer, Vanessa Santiago, and Jeff Falk). The near-coincident positioning of the mic capsules broadens the stereo field but somewhat dilutes the "center-stage" image.

BRIAN KNAVE

SCRATCH HERE

Recording a cappella music may seem low-tech, but there's frequently more to it than meets the ear. Scratch tracks are one example. Because an a cappella group has no band to follow, laying down some reference tracks prior to recording usually helps the group stay in time and in tune.

First, determine the tempo of the song and devise a click track. This can be a straight click, a drum pattern, or whatever works best for the group. However, avoid high-pitched or overly bright sounds, such as claves or cowbells, because these are apt to leak through the headphones and onto the vocal tracks. It may also help to EQ the track, cutting the highs and boosting

the low mids. Even then, it's important to record a bit of the song and listen back to make sure you don't hear leakage from the phones on any of the tracks.

Next, depending on the group and the complexity of the song, you might want to put down some guide chords. These can go at the top of each bar or musical phrase, especially ones leading into difficult chord changes or modulations. A simple bass line might be helpful, too. Use straightforward sounds for these scratch tracks (a piano patch is usually best), and be sure the parts are in time with the click.

Another point: make your reference tracks long enough to reach the end of the song. If they extend beyond the end, though, be ready to fade (or mute) the reference tracks after the last note is sung to prevent sound from bleeding from the headphones.

I usually create the reference tracks on my sequencer (which is synched to my MDMs), both to avoid using up tape tracks and to readily accommodate last-minute changes. Sometimes a group may want to alter the tempo slightly—or even change keys—and it's much easier to make those changes on the sequencer than to rerecord the guide tracks.

For complex, underrehearsed, or long songs, a spoken guide track can save the day. This could include a count-off, location points (for example, "Verse two," "Bridge," or "Chorus"), and even explicit directions for the singers ("Here come the extra two beats!" and so on).

Of course, for a tight, well-rehearsed group, or for one making a quick demo only, reference tracks may not be necessary. You may even find that a click track inhibits the performance, especially if the group isn't used to working with one. In that case, you should dump it. (Obviously, a click is also ill advised if the song contains tempo changes, rubato parts, or other timing anomalies.)

GROUP DYNAMIC

It's possible, of course, to make a solo a cappella record, or for a lone singer to create a group sound via overdubs—a technique Bobby McFerrin has used to great effect on several albums. This article is about recording a cappella *groups*, the members of which usually sing together. Nevertheless, you could approach the recording as you might a band project: recording the rhythm tracks first and overdubbing the solos, the advantage being that the soloists get to keep redoing their parts until they're happy with them.

I find, though, that a cappella groups are usually well rehearsed—much more so than the average rock band—and tend to prefer a "straight, no chaser" approach to recording. Like jazz groups, they rehearse as a unit, with members feeding off the group energy. In such cases, it makes sense to record the group with everyone singing at once, just as you would a concert performance. That's not to espouse a purist approach, nor to suggest that you forego any "studio magic." You might choose, for example, to thicken the sound by doubling some (or all) of the parts. But for me, the joy is in capturing the communication that can happen only when the singers are doing their thing live.

There are three basic miking techniques for recording an ensemble vocal performance: mono, stereo, and multitrack (miking the singers separately). Each approach has its advantages and disadvantages, and there are variations of each. They sound different, too.

ONE-EARED JACK

It may seem primitive, but if done well, mono recording can sound great for a cappella groups. It has tactical advantages, too. First, it requires only one mic and one track (though a nice trick is to double the performance on two separate tracks—leaving any solo parts off of one of the takes—and then pan the tracks apart). Second, it allows the singers to perform without headphones (which not only frees them up but could save you considerable expense). And third, it allows for full visual contact among the singers (see **Fig. 1**), encouraging a tight performance.

For this application, use a large-diaphragm, multipattern condenser mic set to omnidirectional mode. Place the singers at various distances from the mic to "mix" their individual levels, with the lead singer(s) closer to the mic and the booming voices a bit further back. If the singers are wearing headphones, they can move around and determine their own levels. Otherwise, you have to position them.

Omni patterns typically provide accurate and open sound. Provided the room has favorable acoustics, you can make very good recordings this way. You control the amount of ambient sound by moving the singers closer to the mic to reduce ambience or having them step back to increase it. Because the omni pattern is free of proximity effect, you can bring the singers in quite close to the mic without unwanted bass boosting.

If unfavorable room acoustics are spoiling the sound even with the singers close to the mic, try using the figure-8 pattern. This mode has excellent side rejection and relatively small pickup areas, resulting in a tighter, drier sound than omni mode. However, it's hard to fit more than four people (two on each side) into the picture comfortably. As you attempt to add more, off-axis coloration starts to compromise the sound of those not positioned squarely in the pickup areas.

TWO EARS, TWO MICS

The big advantage of stereo recording is that it usually provides the most natural sound—not surprising, considering that we hear with two ears. Also, it takes only two mics, and again, headphones are not required. One disadvantage, though, is that the singers have to face the mics rather than each other, making it difficult for them to see one another. And if they're not wearing headphones, they may have trouble hearing each other, as well.

I'll discuss three stereo-setup options: XY coincident, XY near-coincident, and spaced pair. A matched pair of cardioid condenser mics works best for the first two setups. Cardioids also work for the spaced-pair technique,

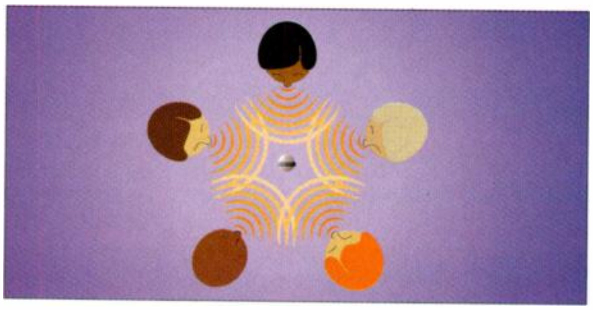


FIG. 1: Even when confined to a single track, you can capture a realistic a cappella performance using a dual-capsule mic in omnidirectional mode.

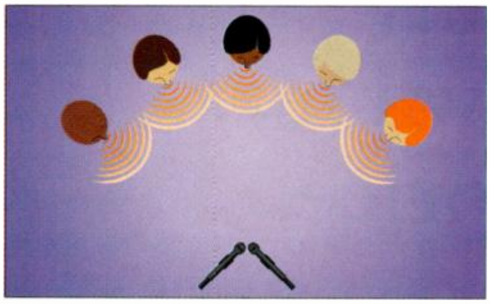


FIG. 2: An XY-coincident pair of mics is excellent for capturing a natural-sounding stereo image.

but I've achieved killer results using small-diaphragm omni condensers as a spaced pair—depending on the suitability of the room acoustics. For all three techniques, stereo sound reproduction is maintained by assigning the two mics to different tracks, which are panned hard left and right during playback and mixdown.

For the XY-coincident pair, position the mics at approximately head height with the capsules at a 90-degree angle to one another (see **Fig. 2**). Then arrange the singers in a semicircle facing the stereo pair, with the lead singer (or singers) in the center and closest to the mics. Done correctly, this technique captures a good stereo spread with precise placement of voices on the soundstage.

For a broader stereo image, try the XY near-coincident technique (see photo on p. 100). However, near-coincident mic positioning can leave a slight "hole" in the center of the soundstage, so don't use it if you want to create a strong center image, say for a single lead voice.

A spaced pair broadens the stereo image further, creating an even bigger hole in the center, especially with cardioid mics. It also increases the likelihood of phase problems, so when using this technique, check the signals for mono compatibility by panning them both to the same spot.

Despite these slight drawbacks, a spaced pair positioned just right can create a desirable sense of spaciousness. This makes it a cool choice for a doubled part, to add dimension to the mix. For instance, you could record the first pass (including the lead voice) with an XY-coincident pair and then use the spaced pair to capture the ensemble doing the same part again, sans lead. Then you would pan all parts hard left and right, and presto: full soundstage—and a nice chorus effect, too.

TO EACH HIS OWN

Recording each singer to a separate track provides by far the most control during mixdown. However, depending on the size of the group, the gear demands may exceed what's available in the personal studio. Each singer requires not only a mic but also a pair of headphones, and probably a headphone extension cord, too. You also need a junction box or headphone distribution amp with enough patches to feed all the phones. Finally, you need enough tracks.

Assuming that you have the gear (and that it's good stuff), the keys to success are mic selection and positioning. Every singer is unique, of course, and there are scores of mics, so it would be unreasonable to make specific model recommendations here. Hopefully, you can make educated guesses based on familiarity with your mic arsenal. From there, it's a matter of trial-and-error until you hit upon the most flattering combinations. (This process can be lengthy for a big group, so you may want to book the testing and tracking sessions for separate days.)

Generally, I use a condenser mic (cardioid, supercardioid, or hypercardioid) on each singer, except sometimes for the bass singer (if I'm going for a particular sound) and the vocal percussionist, if there is one—especially if he or she has worked up a style that includes lots of movement and inadvertent spitting. But mic selection depends chiefly on the type of sound you're going for and how well the mic and voice work together to provide it. Certainly, you can make beautiful-sounding a cappella recordings using dynamic mics alone. What matters most, as with any recording, is the performance.

After determining which mics go to whom, the tracking process is the same as for an individual singer. A pop screen on every mic is a good idea, but if you don't have enough to go around, at least make sure the lead singer has one, as well as the percussionist (that is, if you are using a condenser mic) and anyone else generating excessive plosives. (For more in-depth information about recording vocals, see "Recording Musician: Keeper Vocal Tracks" in the January 1999 **EM**.)

You should have no problem recording all the singers at once in the same room, especially one with good acoustics. Just be sure to use the rear rejection afforded by the mics' polar patterns to create as much isolation between signals as possible. This is usually best done by positioning the singers in a circle (see **Fig. 3**), which also lets them see each other. Some bleed is inevitable, but, kept to a minimum, it shouldn't harm the final mix. If anything, it will make for a more natural-sounding blend.

Some dynamics processing may be called for, especially on the lead voice, which is more likely to span a broad dynamic range. Opt for a transparent compressor (one known not to color the signal much; probably a tubeless, VCA type), and use the lightest setting you can get away with—say, at a ratio somewhere between 1.5:1 and 4:1 and a threshold around -5 dB, depending on the part. The point is to control unruly dynamics while leaving as much dynamic range as possible, to maintain a natural sound. (Compression may also be helpful for the mono and stereo recording applications detailed earlier.)

Although recording each singer to a separate track affords extensive control in the final mix, don't forget that a cappella music is generally best au naturel. Don't go crazy equalizing and processing the tracks; use the control to unify and enhance the overall sound, not to revise it. Assuming the parts are well recorded in a decent-sounding

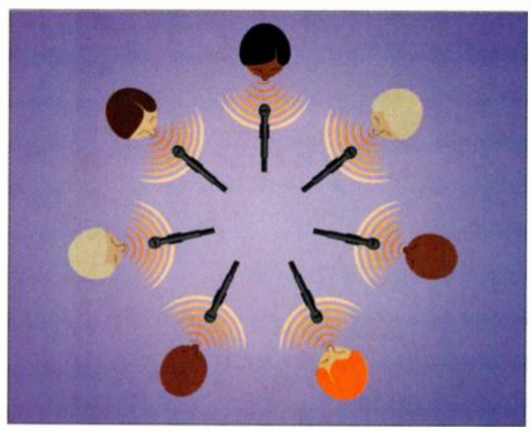


FIG. 3: When recording with multiple mics, arranging the vocalists in a circle helps take advantage of the rear rejection afforded by the directional mics' polar patterns.

room, it should take no more than a few EQ tweaks, a touch of compression, and a bit of reverb to make the final mix shine.

Of course, the naturalist approach doesn't *always* apply. The voice may be the oldest musical instrument, but some a cappella groups today make pretty wild sounds, especially when the voice is used to simulate other instruments. These signals are prime contenders for more extensive tracking or processing techniques.

SHH BOOM BOOM BAP

Vocal percussion is fairly commonplace in a cappella these days, but some groups take the concept to new heights. Check out the Latin group Vocal Sampling, for instance, which makes the sounds of the various percussion instruments typically heard in a salsa band—including guiro, shekere, cowbell, and timbales.

No matter how bizarre the vocal simulation, though, the basic trick is to treat it as you would the actual instrument. For example, for a kick/snare/hats–type groove, try recording the performance in a brighter-sounding, reflective space (a tiled bathroom or stairwell, for example), possibly using additional room mics, to create some ambient attitude. Then, in the mix, pump up the kick with some boosts between 60 and 150 Hz, the "crack" of the snare at 3 to 5 kHz, and possibly the hats with some high-frequency shelving. Finally, to add punch and clarity to the rhythm, squeeze the performance fairly hard with a compressor, say at a ratio between 5:1 and 10:1 with a -10 dB threshold.

THE LOWDOWN

The bass track may also cry out for processing. Say you want an electric bass sound. First, record the voice through a tube preamp to fatten the signal. Next, spend some time with a tweakable compressor exploring different attack and release settings until you find those most complementary to the groove or bass style that you have in mind. Slower settings and moderate ratios (between 2:1 and 4:1) typically work better for slower, ballad-type lines, while faster ones with higher compression ratios suit funkier, popping lines.

Next, equalize the signal to taste, perhaps patching in a sonic enhancer to clarify (or cut) highs and to broaden or tighten the low end. Last, dial in the effects common to the bass style you have in mind—for example, a chorus and big reverb for a Jaco-type fretless sound.

TAKE THE LEAD

One of my favorite vocal simulations is Bobby McFerrin's electric-guitar solo on "Sunshine of Your Love" from the album *Simple Pleasures* (EMI Manhattan Records, 1988). To simulate electric guitar, start with a cheap microphone—one of those Radio Shack models that's designed to plug into a boombox can work wonders, or else try a mic typically used for blues harp, such as a Shure Green Bullet or Astatic JT 30.

Next, run the signal through a guitar rig—for example, a Marshall tube amp with a distortion pedal or wah wah, or whatever is appropriate. Mic the cabinet with a Shure SM57 and compress the heck out of the signal. Then tweak it some more in the final mix, perhaps rotary panning it to create movement. As long as the singer gives you a good guitar simulation to start with, the final effect should be hella convincing.

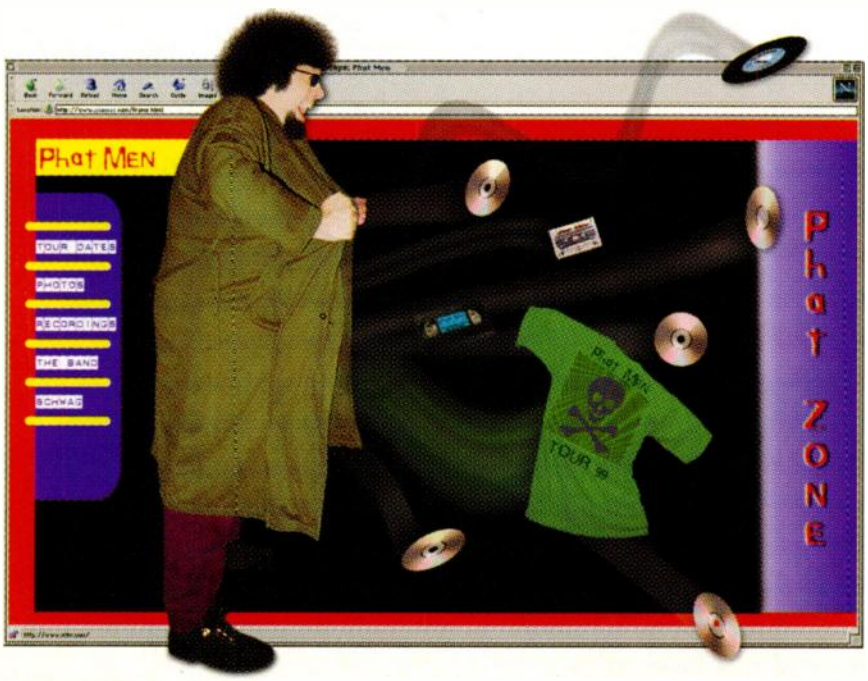
Brian Knave *is an associate editor at* **EM.**

Expose Yourself

Budget-minded tips to make your marketing hip.

By Lygia Ferra and Erik Hawkins

Major record labels spend millions of dollars a year promoting and marketing their acts. As an independent artist, you're lucky if you have any money left to promote your album after paying your recording, mastering, and manufacturing costs. But don't even think about trying to distribute your product without planning a promotional strategy and a budget to go along with it. You need a killer campaign, and you don't have to spend a lot to make it happen. In this article we offer eight proven promotional gimmicks that will help you be seen and heard—and, hopefully, make sales—without costing you an arm and a leg.

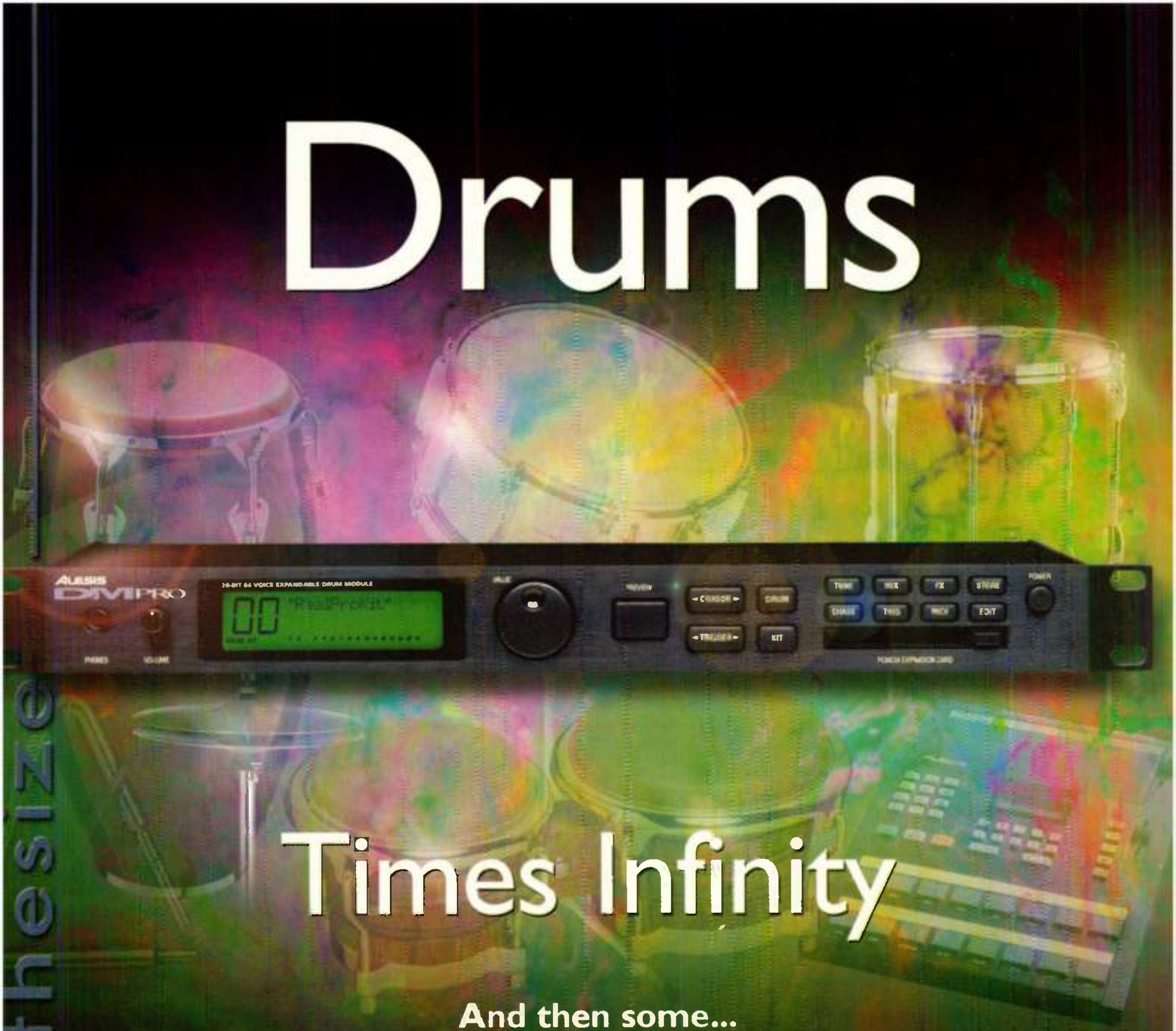


STEVE RAMIREZ

Before you try these tricks, your album must be completed, shrinkwrapped, and ready for sale. Be sure to allow for delays in manufacturing so that you don't waste thousands of dollars on a poorly timed promotion. A campaign budget of $10,000 is realistic, even though that may sound like a lot of money. (For a sample breakdown of what you can do with this sum, see the table "The Budget Laid Bare.") Some people believe that any noteworthy promotion can't be done for less than $20,000, while others have marketed creatively and successfully for around $5,000. Count on campaigning for a minimum of six months; this should be an ample period of time to find out if people are biting. Concentrate on promoting regionally, such as within your state or urban area; targeting a specific location lets you use your limited funds most effectively. Finally, you, or one of your bandmates, must own a computer. You'll need one for everything from tracking sales to printing out consignment contracts to getting online.

SLINGING SAMPLES

Giving away cassette tapes containing two complete songs that represent your musical style gives people a chance to



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The Budget Laid Bare

This table provides an example of how a $10,000 budget can cover a lot of promotional ground. It does not include everything discussed in this article, because not all promo tricks will work for every release. It's up to you to consider your genre and audience and then decide how best to allocate your funds.

Promotion	Description	Time Frame	Cost
Sample Tapes*	2,000 units (clear plastic shells with printing, blank O-cards, no shrink-wrap)	Allow 3 to 4 weeks for duplication and delivery	$1,500
Cable TV Ads	Five or six spots per month	Book 4 weeks in advance	$1,500
Posters	300 two-color 11 x 17–inch posters	Allow a 1- to 2-week turnaround	$400
Print Ads	Five ads per month of various sizes (for example, ¼-page, ½-page, and cooperative ads)	Book 2 to 4 weeks in advance	$2,000
Stickers	1,000 two-color foil stickers	Allow a 1- to 2-week turnaround	$250
T-shirts	300 one-color shirts (225 Beefy-Ts and 75 women's cap-sleeve)	Allow a 1- to 2-week turnaround	$1,275
Web Site*	Domain name and setup fees, six months of service	Allow 2 to 4 weeks for site creation and setup	$275
Postcards*	2,000 full-color, standard-size postcards	Allow a 2-week turnaround	$250
Movie Theater Ads*	A five-screen theater for one month (a total of 2,100 exposures)	Book at least 4 weeks in advance	$800
CD-Release Party*	Caterer and party supplies (including drinks for VIPs)	Plan 4 to 6 weeks ahead (book location, print and send invitations, and so on)	$750
Incidental Expenses	Postage, transportation, phone, and so on		$1,000
TOTAL			$10,000

* To be discussed in the next "Working Musician" column

check out your music without making a cash commitment. Distribute the tapes in places where your potential audience might congregate, such as in cafés, bookstores, or movie theaters, or at venues where artists with similar styles to yours perform. Hand out 1,000 to 2,000 tapes to anybody who will take one. If people like what they hear, great. If they don't, ask them to pass the tape along to somebody who might enjoy it. You really can't go wrong.

Promo tapes don't have to be fancy. A clear cassette stuffed into a blank "O-card" (the cardboard holders that cassette singles often come in) will do the job. You'll save a few cents per unit by not shrink-wrapping, and it will allow you to update information directly on the O-card. The tapes should be just long enough for two songs, one per side. They can be run at high speed (double time); this produces lower quality than real-time duplication, but it costs less and is just fine for sample tapes. Also, have the duplication house print your song titles and contact information directly onto the cassette. This looks more professional than using adhesive labels.

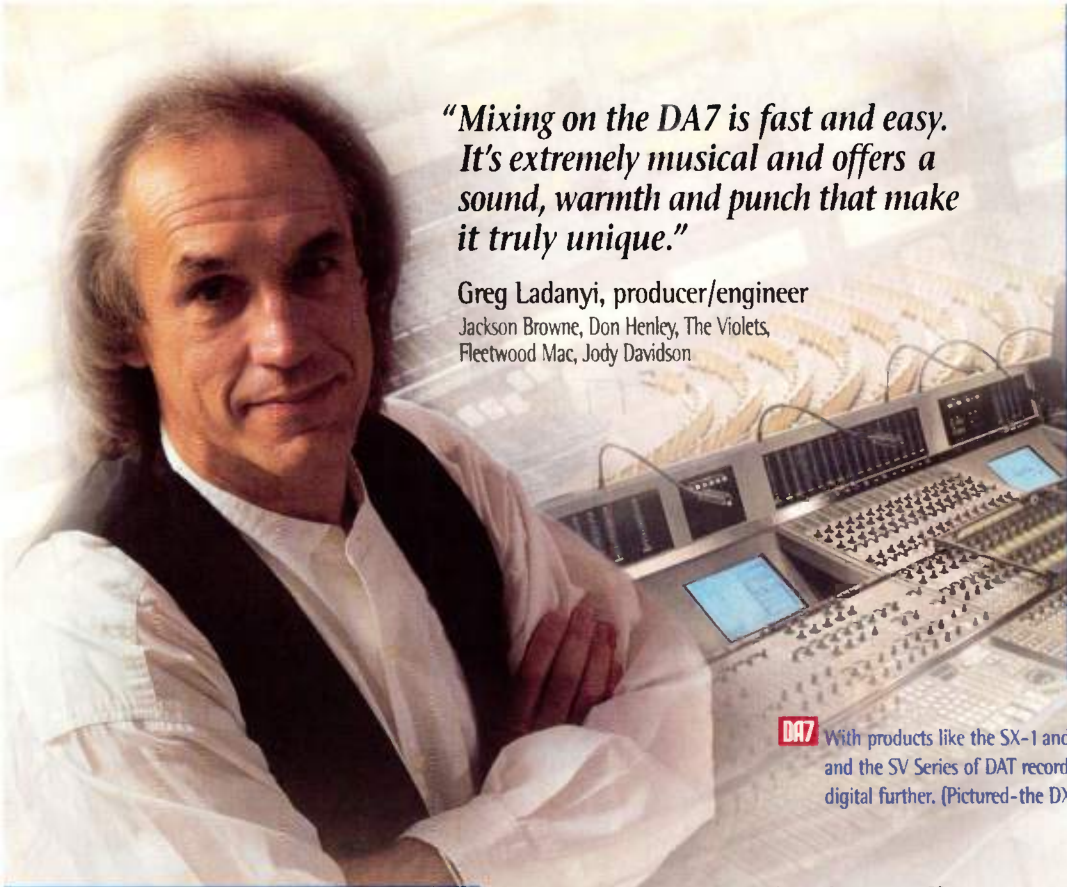
Buy blank O-cards in bulk; if you can't buy them from your duplication house, order them directly through Rainbo Records. (Contact information for resources mentioned in this article is listed in the sidebar "Promotional Resources.") Buy inexpensive inkjet labels at an office-supply store and use your desktop printer to make stickers for the O-cards. Include information about upcoming shows and special offers—for example, "Bring this tape to our next show and get $1 off admission or $2 off our CD" (this way you can recycle your tapes, too). Make another sticker that includes your band's name, the song titles, and your contact information. Figure on spending 70 to 80 cents per cassette. (Prices vary according to tape length and quantity ordered; the more you order at once, the lower the cost per unit.)

There are myriad ways to get an advertising spot produced inexpensively.

YANKING YOUR CABLE

What if I told you that you could get your mug on MTV for less than $100? Local cable companies pipe MTV (as well as other syndicated television programming, such as VH1, BET, Lifetime, and E) into homes throughout the United States. The cable companies sell advertising time to local businesses whose ads then run alongside the syndicated programming. As an independent label, you are a local business and have the right to buy this advertising time, giving you the means to run an ad that appears to be part of MTV programming. (Hint: make your spot look like a music video so that it really fits in.)

Begin by calling your cable-service provider (generally listed under "Cable Television Services" in the local phone directory). Speak with an advertising account representative about buying ad time. A 30-second spot should run between $20 and $75—although depending on the time slot (prime time versus off-peak) and coverage (say, Brooklyn versus small-town Nebraska), you might hear of prices as high as $125 or as low as $15. Keep the demographics of your intended audience in mind. If you're an adult-alternative



"Mixing on the DA7 is fast and easy. It's extremely musical and offers a sound, warmth and punch that make it truly unique."

Greg Ladanyi, producer/engineer
Jackson Browne, Don Henley, The Violets, Fleetwood Mac, Jody Davidson

IF THE DA7 LET YOU WORK ANY FASTER...

DA7 With products like the SX-1 and DX1000 (the DA7's big brothers), and the SV Series of DAT recorders, Panasonic is truly taking digital further. (Pictured-the DX1000 digital console.)

...YOU MIGHT NEED A SEAT BELT.

Let's face it. Today's fast paced production environments require more from a mixer than ever before.

You need instant scene recall, dynamic moving fader automation, 5.1 surround mixing capabilities and an intuitive interface that easily controls analog, digital and MIDI signals. All with 24-bit audio quality.

The Panasonic DA7 digital mixer offers all of this and more. The DA7 has 38 inputs that let you easily route signals between 16 mic/line analog inputs to a hard disc recorder or MDM. You can even get up to 32 channels of analog or 24 channels of digital I/O in ADAT, TDIF or AES format. Or control MIDI devices or sequencer software right from the DA7's 16 MIDI faders.

So if you'd like to speed yourself up without running out of money, take a test drive at your nearest Panasonic Pro Audio Dealer today.

DA7 Many magazine editors have raved about the DA7's exceptional sonic quality, intuitive user interface, automation and affordability. (Some editors even bought the DA7 after the review!)

THE ASSOCIATES

Only $170 a month!

DA7 We've made it affordable. Now you can own a DA7 with our incredible 36-month, no money down, $170/month lease-to-own program.* See your dealer for details or call The Associates at 1-800-526-8638. Make your move to digital now.

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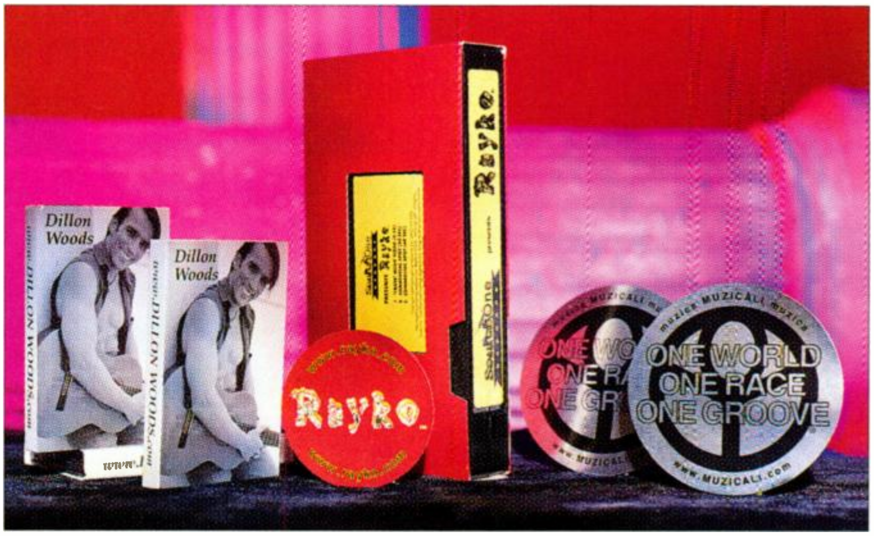

Promotional items such as sample cassettes, stickers, and a short video to use for TV ad spots can be produced cheaply and still get you optimum exposure.

artist, for example, a spot on VH1 would be better than one on BET. Verify with the account rep that you will reach your target market; if not, try a different cable company or a different region.

Making your TV ad doesn't have to be a big production. There are myriad ways to get a spot produced inexpensively: film schools are filled with students looking for exciting projects who will work for free or cost; consumer digital video cameras start at around $900 and give high-quality results; with the right software and hardware, you can now produce video on home computers; and small cable stations might be willing to assemble a spot for you from previous footage (news stories, interviews, live shows, and so on). Using your imagination, you should be able to muster a quality 30-second spot without breaking the bank.

WORTH A THOUSAND WORDS

Billboards are great, but wow, are they expensive. You may be able to get a billboard ad for around $1,000 a month, but in this price range it won't be in a high-traffic area. When pricing billboard ads, ask about "daily effect circulation," or DEC—a figure calculated by a city's Traffic Audit Bureau that reflects the average number of cars passing a particular location each day. If you decide your chosen site has enough traffic to warrant the expense for a month, go for it. But before you do, double-check the area's demographics: it won't help to plaster a country singer's image in a neighborhood where the residents tend to prefer rap music.

Bus-stop ads get you the same kind of exposure as billboards, for a fraction of the cost: about $110 per bench per month. However, bus-stop ad space is usually rented in six-month increments, five benches at a time, bringing the total to about $3,300 for six months. These ads can be a wonderful marketing tool, but again, consider your audience. A rap musician would get more useful publicity from an urban bus-stop campaign than a new-age artist would.

As always, the cost of an ad varies with location. Look in your phone directory under "Advertising/Outdoor" for companies that handle billboards and bus stops in your area. Ask retail stores in the neighborhood if they'll allow you to put up a poster or two in their windows. (Don't consider gluing them up without asking—a vandalism conviction isn't fun.) Once you've created the artwork for your larger-than-life ads, have a print shop turn it into posters. Three hundred full-color 18 × 24–inch posters will usually cost you around $700; two-color 11 × 17–inch posters may run about $400.

ETCH YOUR NAME IN INK

Every major label will tell you that print ads in newspapers and magazines are important promotional vehicles: the amount of money that they spend annually on print campaigns proves this. You probably can't afford to advertise in the same high-profile publications

as a major label, but the alternative news weeklies found in coffee shops and on college campuses will serve your needs nicely. Most of these papers have a decent readership and are regional in scope—just the kind of publicity you're looking for. You can get a directory of these publications from the Association of Alternative Newsweeklies for $15; it covers all the alternative weeklies in North America and includes contact information and demographics for every periodical listed.

A half-page ad in a major alternative paper, such as California's *LA Weekly* or New York's *Village Voice,* will cost around $1,400 a week. Rates are more reasonable for college periodicals (for example, Cornell University's *The Cornell Daily Sun,* which has a readership of almost 19,000). Some college-oriented weeklies, such as Southern California's *Campus Circle,* cover several campuses in a particular region and are even distributed to retail stores and cafés. These publications often offer special discounts for independent vendors. Depending on target readership and region, an ad in a college periodical generally costs about half that of a major alternative weekly. American Passage Media publishes the *College Media Directory,* which lists every college in the country along with its associated periodical and some demographics (but no contact information—you'll have to call the campus directly for that).

Come up with an eye-catching T-shirt design that people will actually wear, advertising your band and your release.

BE COOPERATIVE

Develop a good rapport with radio stations, clubs, and retail stores—and then milk those relationships for all they're worth. A great way to do this is through cooperative advertising. If your music is getting regular rotation on a radio station, ask them if they'll split the price of a print ad with you. The ad can advertise your release and the radio station simultaneously, and will cost you each half the usual price. Splitting costs can also translate into longer running times and bigger ads—benefits for all parties involved. With clubs and retail stores, you might even go beyond print ads and split the cost of cable and radio spots. Cooperative advertising often shows you which entities are just talk and which ones actually put their money where their mouths are.

CHECK ENDORSEMENTS

Artist endorsements are not something that only big stars get. The key is to know which people to approach and how to approach them. Don't even try to get an endorsement deal from a large and well-established manufacturer such as Alesis, Digidesign, or Yamaha. These companies already have endorsees and are inundated with musicians looking for handouts. It's best to approach up-and-coming companies that are hungry for publicity. They want their products to be seen and heard, and that's where you come in. If they like your music, there's no telling what you can accomplish together—from cooperative advertising to product demonstrations featuring your project.

When approaching companies for an endorsement, be persistent, but not a pest. Let them know that you're for real and not just a fly-by-night act. Be patient; they need to feel confident that you're somebody they want to be associated with. (One deal I witnessed was two years in the making, but when the endorsement product was finally handed to the artist, it was worth several thousand dollars.) It also helps to build personal relationships with individuals in the company. Lastly, broaden your efforts beyond just music companies: try clothing designers, shoe manufacturers, cosmetics companies, and jewelry makers. Think like an entrepreneur. After all, you might at least score some products at cost.

STICK 'EM UP

Stickers are inexpensive and can be wonderful promotional tools. Make a sticker that's as much a work of art as it is an advertisement. Nice-looking stickers with a cool design, neat logo, flashy colors, and catchy wording will inspire folks to put them in prominent places. Hand them out everywhere: at gigs, with sample tapes, at parties. You get publicity wherever your sticker gets stuck.

Stickers come in many shapes—from circles and ovals to stars and rectangles—and in a range of sizes and colors, including metallic foils. Visit your local print shop to see what is available. There is even a line of restickable stock that adheres to car windows, mirrors,

PROMOTIONAL RESOURCES

American Passage Media, Inc.
529 Fifth Avenue, 7th Floor
New York, NY 10017
tel. (800) 473-6474 or (212) 622-7330
fax (212) 622-7331
e-mail juliedey@americanpassage.com
Web www.americanpassage.com

Association of Alternative Newsweeklies
1660 L Street NW, Suite 316
Washington, DC 20036
tel. (202) 822-1955
fax (202) 822-0929
e-mail aan@intr.net
Web www.aan.org

Ellen's Silkscreening and Promotional Products
1506 Mission Street
South Pasadena, CA 91030
tel. (888) 545-9711

Rainbo Records
1738 Berkeley Street
Santa Monica, CA 90404
tel. (310) 829-3476
fax (310) 828-8765
e-mail info@rainborecords.com
Web www.rainborec.com

and just about anything else and can be peeled off without leaving any residue.

Create inexpensive photo-ready art for your sticker by using a graphics software package, such as Adobe *Photoshop* or *Freehand,* or *QuarkXPress,* on your personal computer. Costs vary depending on how many colors you use, your ink type, and your sticker stock. Making 1,000 two-color foil stickers, for instance, will cost you around $250.

Create a design for your sticker that you can use for other objects, such as hats, pens, mouse pads, and tattoos. Ellen's Silkscreening and Promotional Products offers to "print your logo on anything"—check out its informative catalogs. Look under "Advertising/Specialties" in your phone directory to find other companies like Ellen's in your immediate area.

THOSE FABULOUS T-SHIRTS

For some reason, T-shirts are heralded as an indicator of how noteworthy a band is. More than one A&R person has commented that receiving a T-shirt from a band makes a good first impression. Most independent artists, however, find it hard enough passing out free tapes, much less free T-shirts. The trick is to give out a few free shirts to influence and impress, and as fodder to *sell* the rest. Give them away to radio program directors, A&R reps, club owners, and press people. Hand out five to ten T-shirts at a live show; when folks see others wearing the shirts, they'll consider purchasing one.

The clincher is having a T-shirt design that people like. It should be as artistic as possible while still advertising your album clearly. The same rule applies as to stickers: a cool logo and design, eye-catching colors, and memorable wording will inspire people to wear your shirts, or at least give them to friends. Either way, they won't be stuffed in a drawer and forgotten.

Look under "T-shirts" in the yellow pages to find silk-screening companies in your area. Prices for silk-screened T-shirts vary greatly depending on the quality of the shirts and the complexity of your design. For example, four-color screening is notoriously expensive, as are designs with lots of fine lines or overlapping colors. Standard crew-neck shirts run from $4 to $8 each, including printing; women's U-neck and V-neck cap-sleeve shirts cost a little more. The more you order at once, the lower the price per unit. For a six-month promotional campaign, 300 shirts is sufficient and cost-effective. Work with the silk-screener to learn what is practical with your design and budget; it's possible to come up with very creative designs using just one or two colors.

STAY TUNED

These eight tips should inspire you to get your promotion plan started. In a future "Working Musician" column, we'll offer other cost-effective ideas for getting independent musicians seen and heard. Until next time, remember: it's not the size of your budget that matters—it's what you can do with it.

Lygia Ferra *is a songwriter/producer whose first independent release as a solo artist,* Strange Peculiar, *is slated to hit retail stores by the end of the year. For more information, visit www.lygiaferra.com.* **Erik Hawkins** *is a musician/producer working in Los Angeles County and the San Francisco Bay Area. He recently started his own indie label, MuziCali Intertainment (www.muzicali.com).*

REVIEWS

E-MU SYSTEMS

PROTEUS 2000

E-mu ups the ante for sound modules once again.

By Gary Eskow

Until E-mu introduced the first Proteus, in 1989, musicians had precious few choices if they wanted a rack-mount sample-playback synth module. Roland had released the U-110 and U-220, and that was pretty much the whole scene. The Proteus/1 changed everything. It offered six audio outputs and 32 voices in a single-rackspace module, and it was 16-part multitimbral. The original Proteus shipped with a whopping 4 MB of 16-bit samples culled from E-mu's Emulator III sound library. They were clean samples, if small and with short loops.

With its ease of use and outstanding capabilities, the Proteus/1 quickly established itself as an invaluable tool for MIDI-based musicians and set the standard for rack-mount, multitimbral sound modules. An entire series of Proteus models followed, mostly featuring new sounds rather than significant architectural enhancements (see the sidebar "**EM** Covers the Proteus").

The Proteus 2000 represents radical rethinking of the Proteus concept, and if E-mu's plans to extend the unit succeed, this could be a revolutionary piece of equipment. I'll say more about those plans later; let's get right to an overview of the unit.

MAKING CONNECTIONS

The Proteus 2000 ships in a basic configuration that users can expand to fit their needs and budget. It offers 128-note polyphony—four times that of the Proteus/1—and has two independent sets of MIDI ports, labeled A

E-mu's Proteus 2000 is a major advance from the original Proteus series, and the manufacturer plans significant further development. The synth module offers extensive programming features, 128-note polyphony, and onboard effects. Its front panel presents a clear view of its parameters.


START AT THE TOP
with the award winning sound, technology, and power, used by the world's most successful professionals, for as little as $49.00 per month.*
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FIG. 1: The Proteus 2000 offers two sets of MIDI ports as well as analog and digital outputs. The digital output duplicates the main outs.

and B (see **Fig. 1**). Section A provides MIDI In, Out, and Thru ports, while section B has MIDI In and Thru jacks. This provides 32-part multitimbral operation, which is great for sequencing. Configuring the unit to receive input from a sequencer on MIDI A and from a keyboard on MIDI B is a breeze and is very handy for live applications.

In keeping with the long-established Proteus routing scheme, the 2000 has six analog outputs: main L/R outputs and L/R submix output pairs 1 and 2. The four sub outputs are actually insert jacks; the tip is the send and the ring is a return that is summed at the main outputs (see **Fig. 2**). This makes it easy to apply an external effects device to Proteus 2000 sounds and mix the processed signals with the unit's internal sounds. A stereo coax S/PDIF digital output duplicates the main outputs.

Fitting the Proteus 2000 into your studio or rig should take no more than a couple minutes. Configure your MIDI patch bay so that it accesses both sets of MIDI connections, hook up the audio ports, and you're on your way.

WHAT'S INSIDE

The Proteus 2000 comes with 32 MB of samples (eight times the capacity of the original Proteus/1) and can be expanded to up to 128 MB. The stock 32 MB includes 1,172 raw Instruments, which are either multisamples or single samples. These Instruments are assigned to Presets, which include all the various programming parameters.

You get 12 banks of 128 Presets each, including 4 banks in RAM and 8 in ROM. User banks 0 through 3 are user RAM Presets and can be overwritten; "composer" banks 4 through 7 store in ROM the same Presets as the user banks; composer banks 0 through 3 contain additional ROM Presets.

Keeping track of more than 1,000 Presets can get confusing, but E-mu's organization scheme is logical and user-friendly. It organizes sounds by raw Instrument type and by Preset category. There are 26 factory Preset categories, and you can create your own, so you can group sounds any way you like.

The sounds in each bank are mostly organized by Preset category. For instance, user bank 1 (composer bank 5) contains five kinds of Presets: patches 0 through 38 are Hybrid/Mixed Keyboards, and programs 39 through 127 are four types of basses, organized in Acoustic Bass, Electric Bass, Sub Bass, and Dance Bass categories. The next bank contains nothing but bass Presets. Other banks contain groups of

For manufacturer information, please see Contact Sheet, p. 179.

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guitars, orchestra hits, brass, and so on.

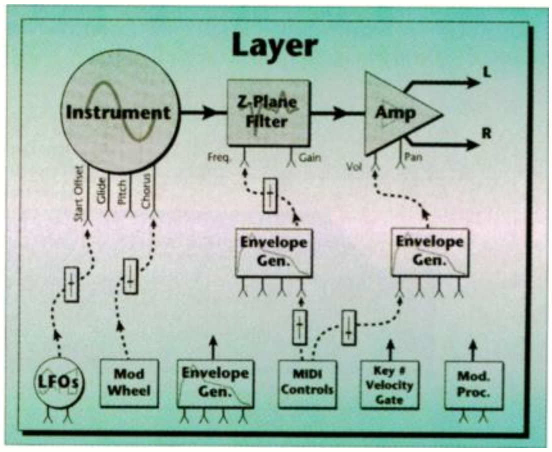
Using E-mu's Sound Navigator feature, you can find a Preset quickly, without knowing what types of Presets are in each bank. You simply change the category field in the main display and move the cursor to the Preset Name field. Now you can scroll through all the Presets in that category. Because you can create your own Preset categories, you can even make custom sets for specific projects. You also can search according to Instrument category, but you cannot create your own Instrument categories.

If you're checking out this box for the first time, you might want to take advantage of the Riff feature. The unit's 387 Riffs are prerecorded phrases that help you preview sounds. Pressing the Audition button on the front panel while you tour the Presets yields a different Riff for each sound—a nice touch that lets you concentrate on the sounds and not your performance. You can assign a Riff to a Preset, but you cannot modify the Riffs.

SOUND JUDGMENT

Generally speaking, I found the Presets to be extremely good. The Moog-style basses, for example, sound authentic on tracks that call for a Moog bass. This is due in part to the samples themselves and to the fact that E-mu has wisely given the user an extensive range of performance parameters to work with. You have substantial control over the way each sound responds to your performing style; you can also customize each sound to the requirements of a track.

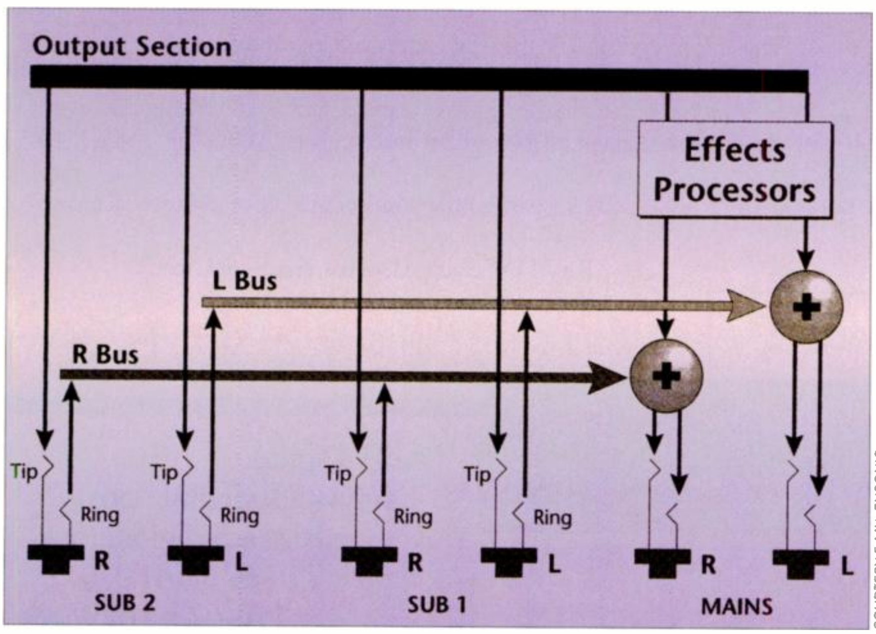
Some sounds are less effective than others, particularly the grand pianos and some of the orchestral instruments. Many of these left me wanting more samples and less stretching, but of course, there is a trade-off at work: the more sounds you have at your fingertips, the less sampling time is available for each. Nevertheless, the Proteus 2000, with its extensive memory, gives plenty of raw sound and power to the user.



COURTESY E-MU-ENSONIQ

FIG. 3: Control sources, shown along the bottom of this diagram, are routed to destinations via PatchCords; up to 24 "patches" can be used per layer.

Many sound modules cry out for editing by way of a patch librarian, but all of the Proteus 2000's many parameters are available from the unit's LCD screen. That's no small achievement. The main window itself takes only a few minutes to understand. In the top left corner you'll see the letter *C*, which stands for the MIDI channel; a number from 1 to 16; and either the letter *A* or *B*. These indicators tell you what channel the sound is on, and in which of the two 16-channel MIDI banks it is. Moving across the top line, you'll see Volume, Pan, and Bank Location information.



COURTESY E-MU-ENSONIQ

FIG. 2: The output section offers six analog outs. Four of the analog outs can be used as effects returns to the main outputs. A stereo digital output duplicates the analog main outputs.

Go down to the bottom line and you'll arrive at the Category column. This is where the Data Entry wheel comes in handy. If you place the cursor on the category, spinning the Data wheel scrolls through the categories. Move the cursor over one (using the clearly marked < and > buttons), and the Data wheel will scroll through all of the sounds within a given category. That's all there is to it!

Many composers use System Exclusive messages to configure their studios before working on a piece. The 2000's Multi-setup feature is well suited to this task. The unit can store up to 128 Multi-setups at one time, each of which saves Preset, Volume, and Pan information for all 32 MIDI channels. Most of the Master menu parameters and a name for each setup are saved, as well. As you would expect, the Master menu parameters include global options such as Bend Range, a Master Tune parameter (in half-step increments), and settings for the 2000's two internal effects processors.

HAVE IT YOUR WAY

The Proteus 2000 offers powerful editing tools, with a structure modeled on the patch-cord paradigm and an extensive real-time MIDI-control interface. Although E-mu has laid things out clearly and does its best to walk the user through the unit's many functions, beginners will no doubt spend a lot of time mastering the intricacies of the 2000's synthesis operations. Thankfully, the manual has a fairly extensive section on synth programming.

Upon entering Edit mode, you use

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the Data Entry wheel to move to each successive screen of parameters. First is the Preset Name screen, which is followed by the Instrument page. Instruments include settings for altering the pitch and start time of the sample. Each of the 2000's Presets can contain up to four layers, with an Instrument assigned to every layer. Other basic parameters for each layer, including an amplifier gain stage that ranges from -96 dB to +10 dB, can also be set in this area.

Moving past the Instrument screen takes you to pages for setting Note and Velocity ranges for each layer in your sound. Then you'll find yourself at the Real-time Crossfade page. I initially flew past this editing page, which was a mistake because it offers extensive control over the way crossfades occur between all of the layers. For instance, I later took a classic synth layer and crossfaded it after several seconds to a percussive piano sample so that only longer notes would have the attack.

Next comes a screen for the Volume Envelope, which is one of three basic envelope generators in the unit. (The other two are Filter and an auxiliary envelope that can be configured to perform a variety of tasks.) The Volume envelope is divided into six stages: Attack 1, Attack 2, Decay 1, Decay 2, Release 1, and Release 2. Like other time-based values, the envelope's time increments use a scale from 1 to 127, rather than absolute time in milliseconds. I've always found this scheme to be rather unintuitive. Nonetheless, all of the envelopes do the jobs they were intended for in a straightforward fashion.

The unit offers 17 types of Z-plane filters. (Z-plane is an advanced filter technology that first appeared in the E-mu Morpheus synthesizer.) Each of the four layers in a Preset can have its own filter, and you can morph between two different filter types using any of the 2000's modulation sources. This adds a lot to the unit's sound-sculpting power.

Composers of electronic dance music, in particular, will appreciate that the 2000's envelopes can be locked to a tempo of your choice using an internal master clock. When this feature is used, the filters can open and close at specific divisions of the beat, in time with your track. You also can lock the tempo-based envelopes to external MIDI Clock.

MORE ON MIDI

The heart of the Proteus 2000's power lies in its real-time MIDI controllers and sophisticated routing system. Using the four Control knobs (or an external sequencer) in conjunction with the Control button, you can access almost all of the box's parameters in real time. (A fifth knob adjusts volume.) By pressing the button repeatedly, you toggle through the three sets of functions that are assigned to each knob. Each of the sets includes four parameters (one per knob), so in total, you can alter up to 12 parameters of a Preset in real time using the 12 customizable controllers.

In effect, the 12 real-time controllers, labeled MIDI A to L, represent connections or "routes" from the knobs or external MIDI continuous control sources to parameters of a Preset. Though all 12 are set by default to control specific parameters, you can redirect them easily. For example, MIDI A (by default, CC 21) is labeled on the front panel as "Tone." Open up the Edit window and you'll see that MIDI A is routed to filter frequency, with an amount that can be set by the user; hence, it's a

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EM COVERS THE PROTEUS

E-mu's Proteus series has steadily expanded since the first module rolled off the assembly line in 1989. Over the years, **EM** has reported on many of these products, mostly in reviews, but also in a major applications feature.

Reviews:

UltraProteus	January 1995
Proteus/3 World	August 1992
Proteus MPS	April 1992
Proteus/2	December 1990
Proteus/1	October 1989

Feature:

The Art and Craft of Using E-mu's Proteus	March 1990

"tone" control. To change the routing, just pick another destination—say, Chorus Amount—and you're all set.

Several control routings are hardwired: Pitch Wheel is always sent to Pitch, and Volume Envelope is always sent to Amplifier, for example. You can set all other connections by using the E-mu PatchCord system. A great deal of real-time control is accessible simply by using the 2000's four knobs and the Control button. If you like to twist and tweak sounds in real time, then you'll appreciate the power these controls give you.

VIRTUAL WIRES

Once you have a handle on how the MIDI controller assignments work, you're ready to move on to the PatchCord part of the 2000's editing scheme (see **Fig. 3**). The 24 "cords" internally connect modulation sources, such as an LFO or white-noise generator, to destinations, which might be filter resonance or chorus amount. You can use up to 24 "patches" per layer.

Most of the Presets that I studied used between 16 and 18 PatchCords to make a sound. Rerouting these cords was very simple. I constructed an ocarina patch and routed Fine Pitch to the Mod Wheel in order to have control over the slight pitch fluctuations that are common to performances on this wind instrument. Used with the real-time MIDI controllers, the PatchCord section offers a vivid palette of colors to use in designing sounds.

GARDEN-VARIETY EFFECTS

The 2000 has a pair of internal, garden-variety stereo effects processors, labeled A and B. Effects processor A produces 44 types of reverbs and delays, while effects processor B generates 32 delay and modulation effects, such as chorus, doubling, flange, pan, and vibrato. The effects are quite serviceable but won't make you want to ditch your external boxes. You can route the processors in parallel and sum their outputs, or you can route the effects in series, with processor B feeding processor A.

There are four internal effects-send buses; you can set a send amount (wet/dry mix) for each sound assigned to a bus. Sends 2 and 3 can also be routed directly to the submix outputs instead of the effects processor, letting you use the sub outputs as send/return inserts for external signal processors.

When you play a single Preset, it feeds the two internal effects processors, and the four sends address the left and right inputs of the two processors. In Multi mode, the effects are global, and you can assign each MIDI channel (that is, each of the synth voices in the multitimbral setup) to one of the four effects sends. This lets you apply a custom amount of either effect (A or B) to each synth voice in the Multi. In this mode, the effects are mono.

MORE IN STORE

Even if these essential features were all the Proteus 2000 offered, the unit would still be an excellent value. However, E-mu has announced plans to integrate the 2000 with its new E4 Ultra line of samplers in an intriguing manner. Remember, the 2000 ships with 32 MB of sound memory, which is loaded into a single slot. The box also ships with three empty slots, which will incorporate sound libraries from E-mu and various third-party sources. (Some of these libraries should be available by the time you read this.) But the most interesting part is that E4 Ultra samplers will be able to burn flash memory that the Proteus 2000 can accept, allowing users to create original sounds and use them as Proteus Presets. That's impressive.

By the way, gaining access to the chip

Proteus 2000 Specifications

Synthesis Type	subtractive sample playback
Polyphony	128 notes
Multitimbral Parts	32
Presets (ROM/user RAM)	1,024 /512
Sample Resolution/ Playback Rate	16-bit linear/44.1 kHz
Effects	(2) 24-bit digital multi-effects processors; 76 total effects types
Controllers	(4) real-time, programmable controller knobs, volume and data knobs
Internal Sound Memory	32 MB (expandable to 128 MB)
Audio Inputs/Outputs	(2) ¼" TRS master L/R outputs; (4) ¼" TRS submix I/O, usable as send-return inserts; (1) coax stereo S/PDIF digital output (AES/EBU compatible); (1) ¼" stereo headphone output
MIDI Ports	(2) MIDI In, (1) MIDI Out, (2) MIDI Thru
Dimensions	1U x 8.5" (D)
Weight	6 lbs., 14 oz.

E-MU SYSTEMS
Proteus 2000 sound module
$995

FEATURES	4.5
EASE OF USE	4.5
QUALITY OF SOUNDS	4
VALUE	4.5

PROS: Generous, expandable sound RAM. Impressive 128-note polyphony and 32-part multitimbral operation. Up to four layers per Preset.
CONS: Numerous synth parameters could be confusing to first-time users. Unexceptional effects.
CIRCLE #437 ON READER SERVICE CARD

set is no big deal. You take off four Phillips-head screws, pop off the top panel, and you're in business.

A NEW GENERATION

E-mu has brought something new to the market with the Proteus 2000, and that's not easy these days. The company has built upon the earlier Proteus sound modules but has added better samples and effects, and much more control over the sounds. I question whether many people will actually spend the time to unlock the power of these tools, given the quality of many of the synth-style sounds already in the unit. But it's good to know the tools are there for the adventurous.

Of course, the quality of the unreleased libraries cannot yet be judged, and as of press time, the plans for integrating this box with the E4 Ultra samplers are just news releases from the manufacturer. But what the Proteus 2000 already offers has to be given very high marks. The unit has eight times the memory of the Proteus/1, four times the polyphony, and twice as many MIDI channels. It's also easily expandable and includes onboard effects, yet it costs the same as the earlier unit. Now *that's* progress!

Gary Eskow *is the New York editor of* Mix *magazine. He is currently producing an album of his solo piano music at Sony Music Studios with pianist Christopher Johnson.*

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CREAMWARE

PULSAR (WIN)

One sweet package with synthesis, I/O, mixing, and effects.

By Allan Metts

When was the last time you had the opportunity to choose how many filters you wanted on your synth, or the number of envelopes you could use when designing a sound? These features and many more are available with CreamWare's new Pulsar DSP sound engine—a single-slot PCI audio card for Windows computers.

The Pulsar is no ordinary sound card. It has 20 discrete channels of audio input and output and enough onboard DSP horsepower to handle real-time mixing and effects. With extensive signal-routing capabilities, MIDI-based control, and excellent onscreen graphics, the Pulsar can easily assume the role of an audio traffic controller in a complex studio.

As an added bonus, Pulsar throws in software synthesis capabilities. Included in the Pulsar package are excellent analog emulations (along with a roll-your-own modular synthesizer), an FM-based synthesizer, sample players that load Akai programs, and even a vocoder. You won't find all the features that exist in some dedicated software synthesis programs, but there is plenty of power to play with. And Pulsar's excellent graphics play a big role in making the synthesizers easy as well as fun to use.

The Pulsar card draws its power from four onboard Analog Devices SHARC DSPs. A cable assembly connects to the card and has MIDI In, Out, and Thru ports (only one of each, unfortunately). Four channels of audio I/O (two channels of stereo analog and two by way of S/PDIF) are provided on the cable assembly through RCA connectors. Pulsar converts digital audio to analog with 24-bit resolution and converts analog to digital at 20 bits. Sample rates up to 96 kHz are supported.

Sixteen channels of audio I/O are provided with two sets of ADAT Optical connectors. These four connectors are mounted directly on the Pulsar card but aren't labeled; I had to dig into the manual to figure out which cable went where. The card has no word clock or 9-pin ADAT sync connectors, so you have to synchronize your ADAT audio using the optical signal itself.

OUT OF THE BOX

I installed the Pulsar into a 400 MHz Pentium II machine with 128 MB of RAM. Hardware, software, and drivers all installed free of incident. I ran the Pulsar software in Windows 95, but it also supports Windows 98. The product does not, however, support Windows NT. (In fact, it might even prevent NT from loading on a dual-boot, dual-processor machine, as one of **EM**'s editors experienced. No other users have reported this problem to CreamWare, though.)

The Pulsar software gives you everything you need to control the card's capabilities in a Windows environment. And like most similar products, Pulsar provides no audio recording or MIDI sequencing functions on its own; you're free to choose your favorite programs for these capabilities. When you launch the Pulsar software, you're presented with a cool-looking application work space (see **Fig. 1**) that contains a Project window and a File Browser. (In Pulsar-speak, a Project represents a complete description of all Pulsar connections, synthesizer settings, and effects.)

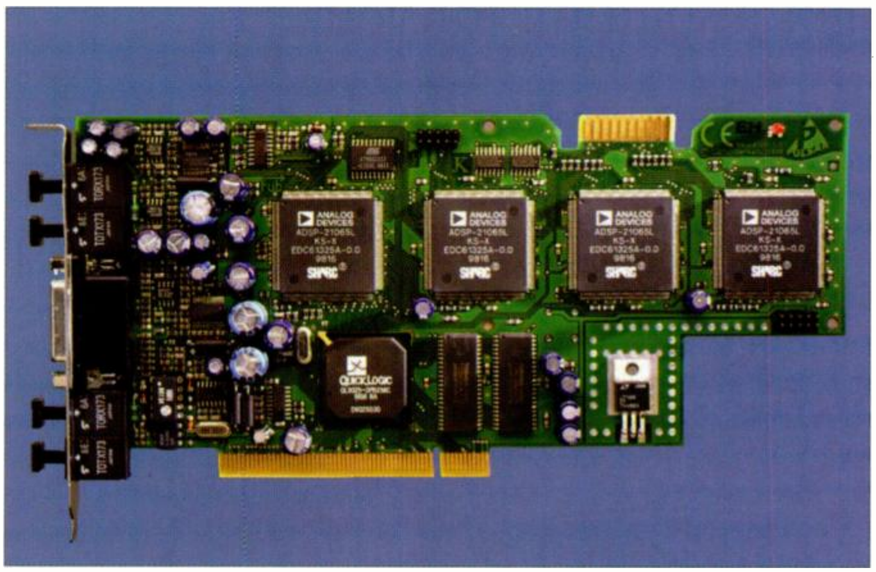
Pulsar's File Browser shows all of the Devices and Modules that you can use in your project. The distinction between Devices and Modules is somewhat blurry. Generally speaking, however, Devices are software synths, mixers, and effects with control surfaces; Modules usually have no control surfaces. The Pulsar's analog outputs and MIDI input are examples of Modules.

CONNECT THE DOTS

The procedure for putting Devices and Modules to work is very intuitive. Choose one or the other in the File Browser, drag it to the Project window, and connect it up. You connect the inputs and outputs of Pulsar objects using "virtual" patch cords. These connections can represent either MIDI or audio data. Each Pulsar Device has a control surface, which is accessed by double-clicking on the Device in the Project window (more about this later).

You can get a different view of your Pulsar connections by opening the Rack window. This window shows each Module and Device stacked one on top of the other as if in a rack (see **Fig. 1**). The inputs and outputs for each object are clearly labeled with a text description that indicates what each Module or Device is connected to. Oddly enough, you don't have the ability to open a Device's control panel from this window.

Other windows in the Pulsar environment include a DSP Load window, which shows how much of your available



CreamWare's Pulsar DSP engine provides numerous audio options and powerful synthesis functions. Sixteen channels of audio I/O are available through the card's ADAT connections, along with two channels of analog and two channels of S/PDIF ins and outs.

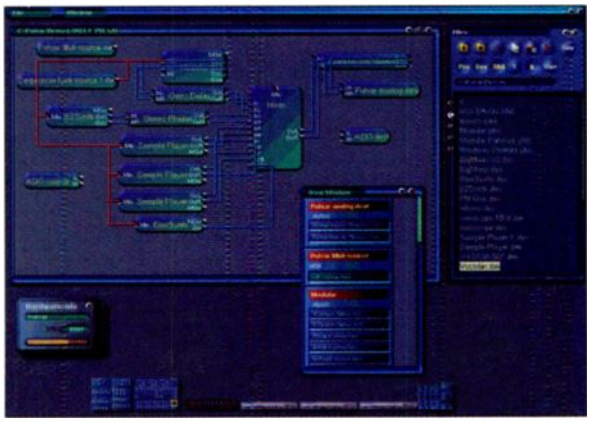


FIG. 1: The Pulsar environment is highly graphical and intuitive. Virtual patch cords are used to connect the many types of Pulsar devices.

processing power is currently in use. A Sample Rate Settings window lets you establish the Pulsar's sample rate and sync source for digital audio. As a master, the Pulsar can operate at 32, 44.1, 48, or 96 kHz. It can also slave to S/PDIF or either of the ADAT inputs.

THE OUTSIDE WORLD

Pulsar is well suited to coexist with other audio and MIDI applications. Each physical connection (MIDI, analog audio, S/PDIF, and ADAT) has corresponding "source" and "destination" modules that can be placed and connected in the Project window.

There are other modules that bring signals into and out of the Pulsar environment but are designed for use *within* your computer. Both 16- and 24-bit Wave modules appear as audio drivers in Windows, which allows you to record Pulsar synth output to your hard drive. First connect the synth to a Wave Destination module in Pulsar, then record the synth by choosing one of the Pulsar's Record sources from within your audio program.

To play back audio "through" the Pulsar, choose one of the Pulsar Play drivers in your audio program. These drivers appear as Wave Sources in Pulsar, so audio is sent directly to one of the card's audio outputs or to an effects module. You can do similar things with MIDI sources and destinations. By default, Pulsar sets up two audio and two MIDI drivers in Windows. You can add more in the Windows Control Panel if you need them.

If you have a particular Pulsar configuration that you'd like to use each time Windows starts up, just set it up and click the Save Project as Windows Standard button. (It's in a Settings dialog box.) This is a nice feature, because it lets you automatically configure the basics for system sounds or simple sequencing without having to fire up the whole Pulsar Project environment. You can also save a default configuration that comes up every time you *do* run the Pulsar environment.

Steinberg *Cubase* users will appreciate the inclusion of 16- and 24-bit ASIO modules, which can be configured to carry 2 to 32 channels of audio. Pulsar also includes modules for sending audio to and from a standard Windows sound card and for accepting audio from programs that use Microsoft DirectSound drivers.

SYNTH-O-RAMA

Pulsar is loaded with software synthesis capabilities, and this is by far my favorite aspect of the Pulsar experience. The program has several analog synths, each of which varies in capabilities and DSP processing load. A few have the look of classic models from days gone by. To work with a synth, open its control surface from the Project window. You are then presented with an impressive display of graphics, knobs, switches, buttons, and faders.

The BlueSynth is one of the more powerful models (see **Fig. 2**). This device has three oscillators per voice; 4-stage envelopes for pitch, filter, and amplitude; and four LFOs that can sync to incoming MIDI clock. (Handy clock dividers are available if you want to group pulses into metrically useful divisions.) Velocity, Aftertouch, and the LFOs can modulate the filter, the amp, or any oscillator.

Although the BlueSynth is useful, it lacks portamento or a noise generator. If you want these things, however, check out the miniScope. This synthesizer, which looks suspiciously like a Minimoog, has three oscillators per voice, a noise generator, and six LFOs.

Pulsar also includes EZSynth, which has only a single oscillator and filter; distortion and chorus are available to fatten up the sound a bit. EZSynth is a good choice when you want to use a minimum amount of DSP power. Another option, Inferno, is also a one-voice synth, but it provides a more capable oscillator (including a fixed-ratio suboscillator) and more control over the amplitude and filter envelopes than EZSynth. Like EZSynth, Inferno includes chorus and distortion and also offers a ring modulator and Velocity sensitivity.

U KNOW 007, presumably an emulation of a Roland Juno-series synth, has a sawtooth oscillator, a pulse oscillator, a square-wave suboscillator, and a noise generator. U KNOW 007 also gives you a high degree of control over the phase relationships between the two main oscillators. The synth includes niceties such as a spread (detuning) control, a highpass filter, delayed onset of LFO modulation, stereo chorus, and an ability to link the amplitude and filter envelopes.

ALTERNATE MODES

If analog emulations aren't your bag, Pulsar offers several alternate functions. Among these are the Sample Players, which can layer up to four Akai-format programs. The Sample Players have only minimal editing capabilities (one model offers transpose and amplitude envelope offsets; the other adds filter offsets as well), so your best bet is to create your Akai programs

FIG. 2: The BlueSynth is a three-oscillator model, with plenty of control over filter, envelope, and LFO settings.

For manufacturer information, please see Contact Sheet, p. 179.

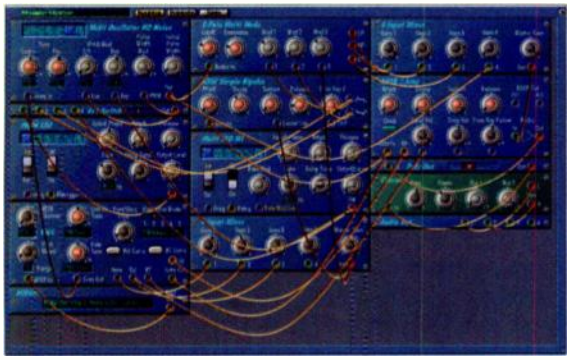


FIG. 3: A modular synthesizer providing 80 synthesis modules is included with the system. Synthesizers are built by connecting any number of arbitrary modules.

elsewhere, or choose from a variety of sounds on the included sample CD.

Another module is a monophonic FM synthesizer, with lots of graphical parameter screens for editing waveforms, envelopes, and operator algorithms. I don't have much use for the sounds that monophonic FM synthesis can create, but Pulsar's intuitive user interface provides an excellent medium for learning and experimenting with this complex method of synthesis.

If none of the ready-made Pulsar synths suit your needs, then you'll appreciate the ability to "roll your own" modular synthesizer (see **Fig. 3**). There are 80 synth modules in all, and each can be added multiple times to your onscreen modular creation. You'll also find plenty of oscillators (including two wavetable oscillators from Waldorf), filters, envelopes, and LFOs. Rounding out the toolkit are audio mixers, signal switches, MIDI manipulators, and effects.

To create a modular synth, add the modules you want and patch them together. Each module comes with a variety of ports, which might carry frequency, gate, or envelope signals. Others offer MIDI, audio, or a DC offset. Two clicks of the mouse is all it takes to place a patch cord from one port to another. I routed signals and twiddled knobs for hours and came up with all sorts of interesting sounds.

All Pulsar synths are monophonic by default, but that doesn't mean they have to stay that way. With the exception of FM One, each synthesizer can operate with up to 16 notes of polyphony. (I found it interesting that you *can* do polyphonic FM synthesis with the modular synth.) And if that's not enough, just add more copies of the synthesizer to your Project work space. More polyphony carries a cost of greater DSP consumption (as does synthesizer complexity), so there are limits to what you can do. I was able to easily create a 16-note EZSynth, but I ran out of DSP power when my U KNOW 007 synth exceeded 13 notes.

EFFECTIONATELY YOURS

Pulsar includes a decent set of effects, which you hook up in the Project window. Included in the effects palette are delay, chorus, flanging, phasing, and dynamics processing. You also get a 4-band parametric EQ, a 4-pole filter (with a built-in LFO), stereo chorus, and a nifty 11-band vocoder. No reverb is offered currently, but one is in the works.

With the exception of the vocoder, the effects are divided into aux and insert effects. Both types can be hooked up in the Project window, but insert effects can also be dropped into an effects slot on the Big Mixer (described shortly). Most inserts are mono, but stereo versions of the compressor, limiter, and 4-pole filter are available.

The aux effects include stereo versions of delay, chorus, flanging, and phasing. Everything but chorus is also available as a mono insert effect, and in a "cross" version that applies the left channel effects to the right channel output and vice versa. All of the aux effects can operate in a mono-in/stereo-out configuration.

The vocoder stands on its own and has audio inputs for analysis and synthesis. It includes a built-in frequency analyzer and all the tools you need to map 11 bands of input frequency to the output signal. Move over, Alan Parsons!

All of the effects have graphical control surfaces and are easy to use. I particularly like the dynamics processing (see **Fig. 4**), which shows a gain reduction curve and real-time meters for input, output, and gain reduction. You can adjust the effect's settings by turning knobs, by entering numbers directly, or by manipulating the gain-reduction curve with the mouse.

All of the effects sound great to me, and I noticed no unpleasant artifacts during the course of the review. I particularly enjoyed making low-budget-sci-fi sounds with the 4-pole filter. (I'm easily entertained.)

MIX IT UP

With so many places for audio to come from and go to, you definitely need mixing capabilities, and Pulsar gives you two mixers to choose from. The Dynamixer can be configured to accept 1 to 16 inputs and offers bare-bones mixing (as well as bare-bones DSP consumption) to a stereo output. Each input channel has gain, pan, and mute controls, as well as a channel fader and meter. Separate, linkable faders and meters are provided for the two output channels.

If you need more mixing flexibility, bring in the Big Mixer (see **Fig. 5**). This is a 32 × 16 × 2 configuration with dedicated outputs and submix strips for the 16 buses. Each of the 32 input channels has a gain control, a phase-inversion switch, a separate stereo send to the monitor bus, six mono aux sends, and up to four insert effects.

Mute, solo, pan, prefader listen, and a 4-band parametric EQ round out the offerings for each channel strip. Each of the sends, EQ, and effects can be configured to operate pre- or postfader, and each channel's level meter can monitor the input or channel level. You can assign each channel to one of six mute groups and to 2 of the 16 buses. Or, if you prefer, you can send the

FIG 4: Pulsar has several useful effects, including the compressor shown here. Notice the input/output curve and real-time meters.

Pulsar
Minimum System Requirements
Pentium II/300; 128 MB RAM; Windows 95/98; 16-bit sound card

channel directly to the main stereo bus.

The Big Mixer provides stereo returns for each aux send. You connect effects to the aux buses by patching Pulsar effects in the Project window. External effects can also be used by connecting the Pulsar's audio ins and outs to these buses instead. The Big Mixer has master aux send and return controls, and each return can be fed into buses 1 through 16, the main mix, or the monitor bus.

Insert effects get dropped into a channel strip from the File Browser, and from there you can open the effect's control surface to adjust it. You can also drop up to four insert effects onto the master output section.

Speaking of which, the master section has linkable faders for the left and right channels, as well as separate level controls for the control-room and monitor outputs. The control-room outputs can contain the main mix, the monitor mix, or the prefader listen signals. A convert-to-mono switch, a 4-band EQ, and a talkback circuit round out the master section. A Big Mixer, indeed!

MORE CONTROL

Every Pulsar device has an associated Presets list, which lets you store a snapshot of each device setting. Presets are particularly important for the synthesizers, because they let you get back to a particular sound with ease. But presets are also useful for quickly changing the configuration of a mixer or effects device. You can export preset lists to separate files, and thereby keep different sets of presets for particular projects.

Most of the synthesizers come with a ready-made collection of useful presets (synth patches, for example). Unfortunately, none of the effects has such a collection. I'd like to see a set of commonly used settings for each of the effects devices.

Pulsar offers extensive MIDI control over the synth and mixer settings. A simple right-click on a Pulsar control is all it takes to open a MIDI controller assignment window. Once there, Pulsar can autodetect incoming MIDI messages to make the controller assignment a breeze.

Pulsar's MIDI controller assignments have some powerful features. You can

CREAMWARE
Pulsar DSP engine
$1,298

FEATURES	4
EASE OF USE	4
AUDIO QUALITY	4.5
VALUE	4.5

PROS: Impressive software synthesis capabilities, with real-time mixing, I/O, and effects. Highly intuitive onscreen graphics.
CONS: No Windows NT support. No reverb. Nonstandard Windows user interface. Occasional program crashes.
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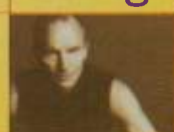


Les Paul Award

FIG. 5: Pulsar's Big Mixer provides 32 inputs and offers 16 buses along with effects sends, a monitor mix, and even a talkback circuit.

automate entire mixes by linking each fader in a mixer to the MIDI stream from a sequencer. You can control fat filter sweeps from a breath controller. I only wish that Pulsar's effects had the same capabilities. Having external control over delay time or chorus depth would be nice as well.

PULSE CHECK

That the Pulsar system offers an extensive set of features should now be obvious. I should note, however, that you can't use all of the features simultaneously. The Pulsar card provides a fixed amount of DSP processing power, so you have to choose among the available Pulsar devices.

So just how much power is there? One of the Pulsar demo Projects uses most of the DSP power, so I'll describe it here as a representative example. This particular project has a BlueSynth, a Modular Synth, an EZSynth, and three sample players. The six sound generators all feed from the same MIDI source, and between them they contain 11 notes of polyphony.

One synth feeds into a stereo delay, and another feeds into a stereo phaser. Everything goes into an 11-channel Dynamixer. The output of the Dynamixer goes through a stereo compressor on the way to the analog outputs. As you can see, a significant amount of Pulsar's capabilities can be used at the same time.

The question of latency always comes up in discussions of software-based synthesizers. I'm happy to report that Pulsar performs admirably in this regard. Notes sounded as soon as they were played, and the effects of moving an onscreen slider or knob could be heard in real time. I did notice a tiny delay when playing a piano program with a Pulsar sample player, but this wasn't a problem for me.

Pulsar's graphics are impressive, but applications that replace or remove standard Windows functionality always bother me. Pulsar's windows can't be resized from the top, which makes it impossible to shrink a window that extends below the bottom of the screen. Other annoyances include menus that don't close when they should, cursors that don't change shape when they ought to, and scrollbars that can't be moved in certain ways. The main application work space can't be resized with the mouse—you have to open a dialog box to do it.

The Pulsar environment is stable, for the most part. I did, however, experience a few nonrepeatable Pulsar crashes during the review period, and these usually meant I had to restart Windows before I was up and running again. I also found some user-interface quirks, which indicate that Pulsar isn't completely polished yet. But this technology is new and very complex, so a few quirks are understandable for now. Overall, I was satisfied with the performance of the product.

Most of the documentation is onscreen in PDF format, which is undesirable for those who like printed manuals. Nonetheless, the documentation is thorough and well organized. Unfortunately, no context-sensitive help is available.

Overall, this is a great product and a great platform. Where else can you get lots of software synthesis, mixing, I/O, and effects for under $1,300? The sound quality is excellent, and the product's usability is superb. If you want to add some fat analog emulations to your sonic arsenal, or if you'd just like to add some DSP horsepower to your PC, you should give Pulsar a thorough test-drive.

Allan Metts *is an Atlanta-based musician, software/systems designer, and consultant.*

SHURE BROTHERS

KSM32

A smooth-sounding studio mic raises the standards within its price range.

By Myles Boisen

The venerable Shure Brothers, long known for manufacturing affordable and durable "working class" mics such as the SM55 (or the "Elvis mic"), SM57, SM58, and SM81, has made a bold move with the creation of the company's first new studio condenser microphone in years. At a time when many manufacturers are struggling to offer scaled-down versions of their premium mics to the personal-studio market, Shure has gone upscale to meet the competition with the KSM32. A fixed-cardioid, electret condenser mic with a ¾-inch diaphragm, the KSM32 boasts smooth sound, sleek styling, and a lavish feature set that puts it on a par with the multitude of prestige imports.

BEAUTY PLUS

The KSM32's tapered body gives it a sleek, distinctive look. I reviewed a pair of the more expensive KSM32/SLs, which sport a lustrous champagne finish and ship with a full complement of deluxe accessories. (The less expensive KSM32/CG has a charcoal gray finish and comes with a swivel mount and padded carrying bag.) All the exterior parts of the microphone are metal, precisely machined, and built to last. Even the triple-layer mesh grille surrounding the side-address capsule is unusually sturdy and refused to change shape under considerable pressure.

A modern Shure logo identifies the front of the KSM32; as a nod to the retro crowd, the rounded, old-style, 1950s Shure logo is stamped on the back. Also on the back of the microphone is an accessible 15 dB pad switch and a highpass filter with three positions that provides a flat response with either an 18 dB/octave cut at 80 Hz, or a 6 dB/octave cut at 115 Hz (see **Fig. 1**). Other nice touches include gold-plated XLR connector pins, curved ribs that sweep up the side of the mic to make its satiny surface easier to grasp, and no visible screws or mounting threads to disrupt the mic's elegant lines.

Electronically, the KSM32 raises the ante for its price class, with Class A preamplifier circuitry, a transformerless output, and an innovative capsule design. By using a low-mass ¾-inch diaphragm that is embossed with diamond-shaped bumps to increase its surface area, Shure has provided the exemplary transient response that small-diaphragm mics are known for, along with the extended low-end reproduction more commonly heard from large-diaphragm mics. The capsule backplate is permanently charged (that is, the mic is an electret), and Shure claims to have solved the long-term reliability problems that commonly plague inexpensive electrets. (Just the same, the company avoids using the term *electret* in the user manual.)

The KSM32 capsule and preamp are designed and built entirely in the United States by Shure Brothers, and the primary components are assembled at the Shure plant in Evanston, Illinois. (Shure doesn't identify the country of origin for the parts.) Shure's stringent standards are evident in the KSM32's superior construction quality.

SHOCK VALUE

The KSM32/SL comes with a cute, lunchbox-size flight case decorated with Shure's retro S logo. With tough plastic corners, braced hinges, locking latches, and multiple layers of protective foam inside, the case far surpasses what one would expect to receive as standard issue accompanying a mid-priced mic. And the case is stylish, too. Unfortunately, its aluminum sides are easily scratched and scuffed with everyday use. So if you like to keep your toys in showroom condition, you may want to use the case for studio storage and transport the KSM32 in its nifty red velveteen pouch.

Shure generously includes a suspension shock-mount, which encloses the lower half of the KSM32 in a teacup-size basket and effortlessly guides the mic base into a threaded ring that screws on flush with the bottom of the XLR jack. The basket is made of plastic, which ensures that it won't scratch the mic, and appears to be unbreakable. Thick elastic bands anchor the basket to an outer ring. An adjustable, locking swivel-mount secures the assembly to a mic stand. While penny-pinching competitors contribute to future landfills with cheap plastic "hardware," Shure uses all-metal parts for the threaded fittings and bulk of the shock-mount.

In addition to the deluxe shock-mount (which works beautifully), Shure also provides an all-metal swivel-mount, for those occasions when the capsule's internal shock-mounting system is sufficient or when the shock-mount is too unwieldy. A brief, but thorough, user guide is also included.

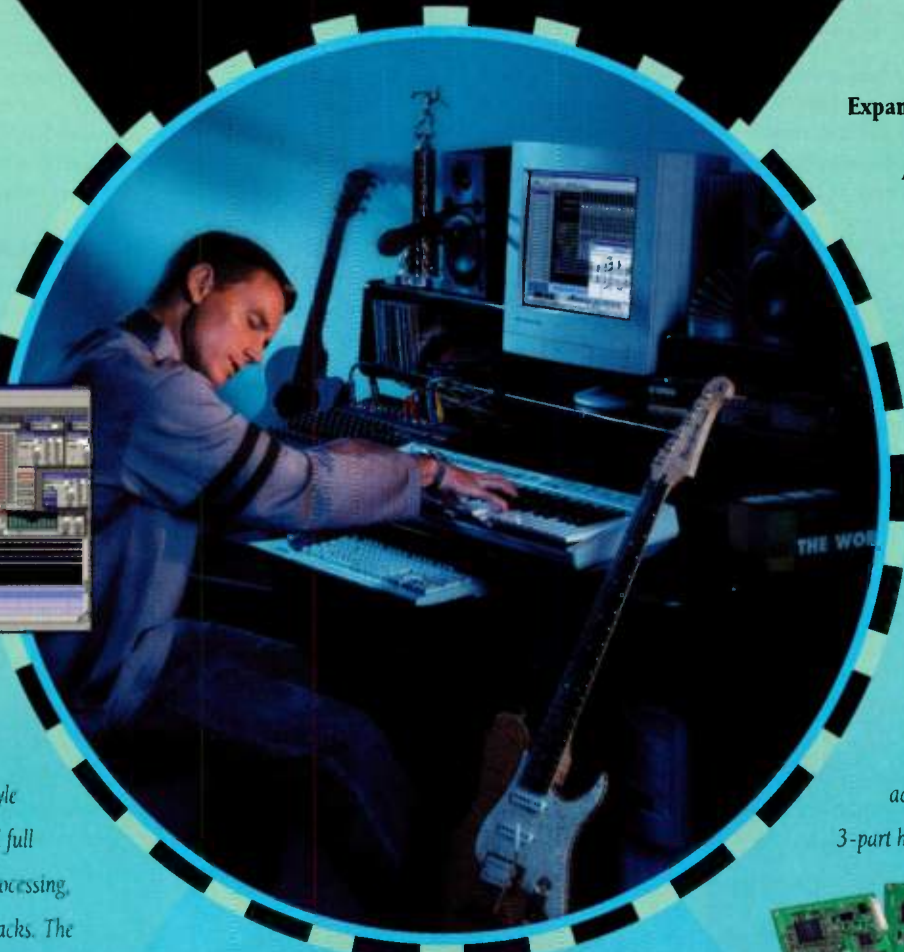
DRUM DUTIES

I first tried out a pair of KSM32s as drum overheads on a session for the band Slumber Inc. The Shure pair compared favorably in low-end response with my usual first-choice drum mics (a matched pair of cardioid Oktava 012 small-diaphragm condensers), but the KSM32s' mellow high end was unable to deliver the measure of extra brightness that drummer Richard Colbert needed to cut through this band's raucous wall of guitars.

I got better results by splitting up the pair on the Slumber Inc. drum kit,

Every detail of the KSM32 makes clear that Shure Brothers took its time and did its homework before releasing this fixed-cardioid condenser mic. No look-alike or sound-alike, the KSM32 is a versatile studio microphone with smooth highs, accurate bass reproduction, and exceptional transient response.

You could spend years putting together a powerful, fully equipped MIDI/Audio project studio.



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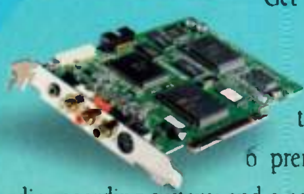

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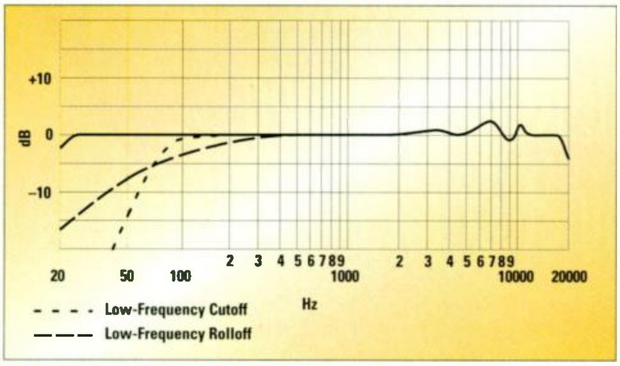

FIG. 1: Shure's frequency-response graph reveals an even bass response and an uneven top end, with a 3 or 4 dB boost centered at 7 kHz, a sharp dip at 9 kHz, and another sharp peak just above 10 kHz. Applying the mic's highpass filter results in an 18 dB/octave cut at 80 Hz or a 6 dB/octave cut at 115 Hz.

using one KSM32 for drum-room ambient miking, and the other on hi-hat. Both tracks not only sounded great but were ready to mix without EQ.

On this session, the KSM32 was ideally suited for use as a close mic for the acoustic strumming sound on the group's solid-body electric guitars. I have used this technique of combining amp sound and plectrum strumming with a variety of guitars and mics, and in this case, the KSM32's small diaphragm was able to grab the fast-transient pick sound that I wanted without a hint of brittleness.

My studio partner, Bart Thurber—a San Francisco Bay Area recording veteran who has worked with literally hundreds of punk, garage, and alternative bands—put the two mics to the test on several occasions, usually as a spaced overhead pair on hard-hitting drummers. He was full of praise for the KSM32's hot output, low noise, tonal balance, and pleasing accuracy in our live drum room, where he sometimes mounted the mics upside down (with the front of the mics pointed up at the ceiling, rather than down at the drums) as an effective way of dealing with overbearing cymbal bashers. Thurber rated the KSM32 as "especially good" on percussion.

When used as a spaced pair on a set of three conga drums, the KSM32s gave an honest, well-blended sound and displayed surprisingly good rejection of a drum set located only three feet away in the same small room. The Shure pair wasn't quite as bright as the mics I normally use for this purpose, but they also sounded less hyped and more true in the highs. When the conga player confirmed that there wasn't enough "slap" on his tracks to cut through the dense Latin-jazz mix, a conservative high-end boost was sufficient to bring out the skin sound to everyone's satisfaction.

WHIRLING GLADNESS

I also had occasion to use the KSM32s on a Leslie 120 rotating-speaker cabinet. Here, the pair yielded excellent presence and tight stereo imaging, while softening the cabinet's slightly distorted, scratchy highs. In a way, the KSM32 worked as a good ribbon mic would in this application, downplaying the top end and bringing out an

For manufacturer information, please see Contact Sheet, p. 179.


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SHURE BROTHERS
KSM32 condenser microphone
KSM32/SL $1,029
KSM32/CG $959

AUDIO QUALITY ■■■■
VALUE ■■■■■
1 2 3 4 5

PROS: Extended, accurate lows and rich mid-range. Smooth, natural high-end response. Exemplary transient response. Tight cardioid pattern. Low self-noise. Excellent resolution, depth, and dimensionality. Class A electronics with transformerless output. Durable metal parts. Metal swivel-mount. Demo units sufficiently matched for non-critical use as a stereo pair.

CONS: Raspy and uneven narrow-band resonances above 6 kHz are unflattering to some vocals. Prominent midrange, sometimes sounds murky.

abundance of rich midrange tone. Once the band members heard the results, they wanted to put the stereo-miked Leslie on every song!

VOCAL PEAKS

Based on its performance as a natural-sounding, detailed, and fairly warm transducer, the KSM32 seemed a good candidate for miking vocals. But after using it on three singers (two male and one female), I had some reservations about the mic's otherwise laudable high-end qualities. To my ears, all these diverse vocalists—a blues singer, a spoken-word artist, and a mariachi bandleader—sounded uncharacteristically "buzzy" on the KSM32 (recorded at a standard working distance of six to nine inches from the mic capsule).

This is a matter of personal taste to some extent; in fact, one of the singers preferred the KSM32's sharp-edged sound over that of a favorite tube mic I had used to record the performance simultaneously to another track for comparison. But I was bothered by what I perceived as uneven and raspy-sounding narrow-band resonances above 6 kHz. Almost any worthwhile vocal mic will exhibit some presence boost; however, models optimized for this purpose tend to incorporate more gently rising, shelving-type boosts, which augment broad regions of a singer's high-overtone range.

On the other hand, the KSM32 lived up to the even bass response shown in the manufacturer's frequency-response chart (see **Fig. 1**), supporting the male vocalists' low ranges without undue clouding or proximity-effect bass boosting. The frequency-response chart also confirmed my impression of the KSM32's uneven top end, showing a 3 or 4 dB boost centered at 7 kHz, a sharp dip at 9 kHz, and another sharp peak just above 10 kHz.

AN IMPERFECT MATCH

According to a Shure Brothers representative, the response graph was obtained by averaging the output of 100 KSM32 microphones. Company literature states, however, that the manufacturer "has no formal process to match pairs of microphones," and it is unreasonable to expect mass-produced mics such as these to equal the consistency of higher-priced European mics designed for critical stereo recording. Indeed, when listening closely, I detected an audible difference in the general high-end response of the two review mics.

To confirm this, I set up a loudspeaker steady-tone test for frequency matching of the two mics and then documented minor (2 to 3 dB) mismatches at 800 Hz, 11 kHz, 12 kHz, and 17 kHz. At 16 kHz, I detected a 4 dB difference in response between the two mics. A variation of 2 dB or more from the published chart could explain the anomalous upper-range buzziness I heard.

In a sweep-tone test, the KSM32 pair again showed a high degree of variation above 10 kHz, but below that frequency, they appeared well matched. Overall, although not a perfect match, the two KSM32 mics were sufficient as a stereo pair for general pop-music use.

CABINET RESERVATIONS

The only other disappointment I had was when I placed the KSM32 on a cranked-up Bogner guitar cabinet and found it to sound surprisingly distant and dull at just ten inches away. Normally, this "not too close, not too far" placement works well to capture an amp's cutting high end while adding a bit of room ambience to the track. But in this case, the mic just sounded mushy, possibly because the 45-degree angling relative to the cabinet picked up unfavorable off-axis midrange coloration.

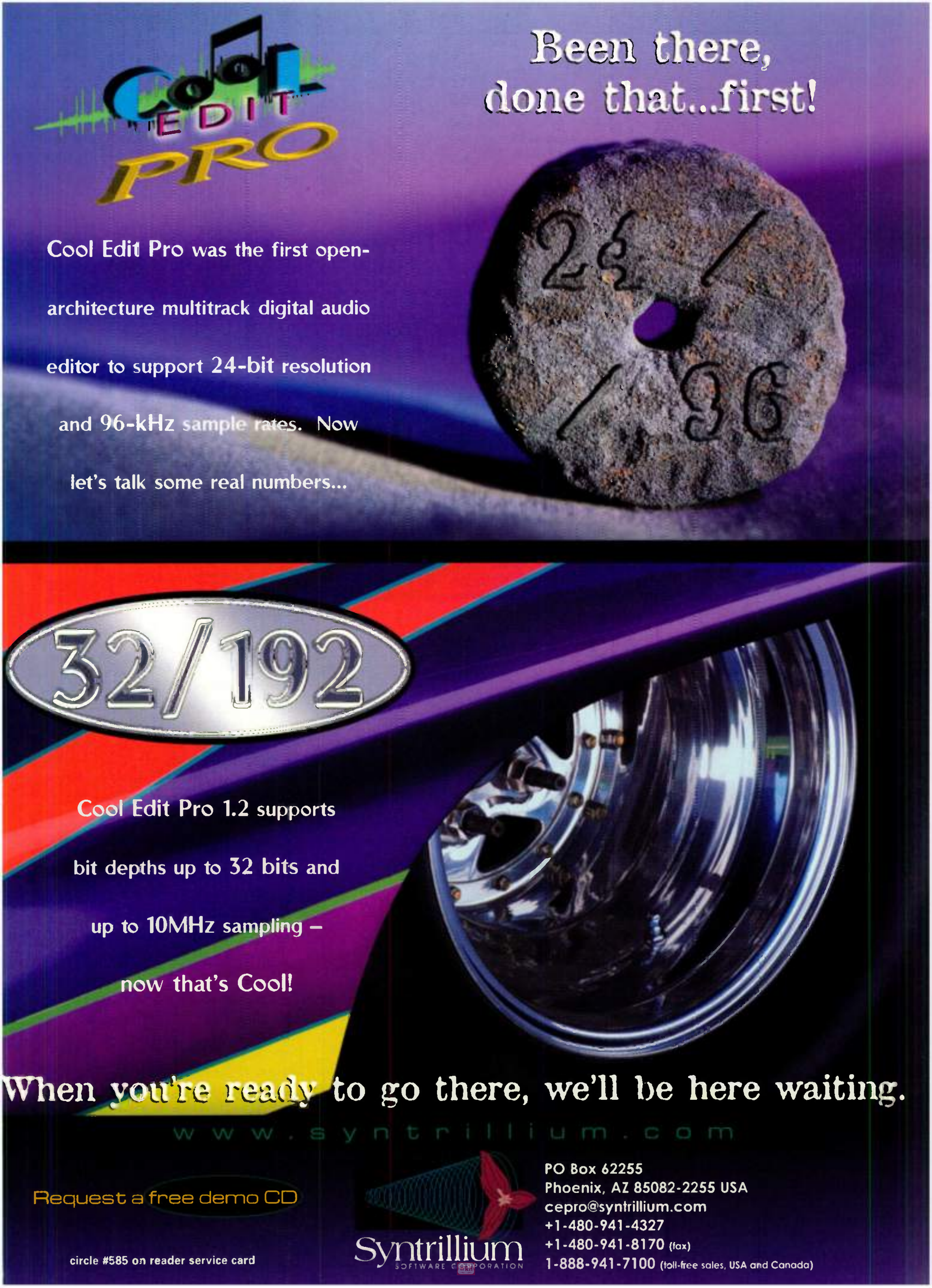
In all other applications, though, I found the KSM32's tight cardioid pattern (which, as with most directional mics, varies from subcardioid on bass frequencies to supercardioid above 10 kHz) to be an asset.

SERIOUS TOOL

To gain further insight into the Shure's personality, I compared it to a couple other microphones using recorded music as the sound source. In these

KSM32 Specifications

Element	fixed-charge backplate, permanently polarized (electret) condenser
Diaphragm	¾", 2.5 micron, embossed gold-vapor-deposited Mylar
Polar Pattern	cardioid
Frequency Range	20 Hz–20 kHz
Dynamic Range	126 dB
Signal-to-Noise Ratio	81 dB
THD (20 Hz–20 kHz)	<0.08% (@ 120 dB SPL input)
Sensitivity	-36 dBV/Pa (16 mV)
Self-Noise	13 dB (A weighted)
Maximum SPL	139 dB
Highpass Filters	80 Hz, -18 dB/octave; 115 Hz, -6 dB/octave
Attenuation Pad	15 dB
Dimensions	7.4" (L) x 2.2" (D)
Weight	1 lb., 1 oz.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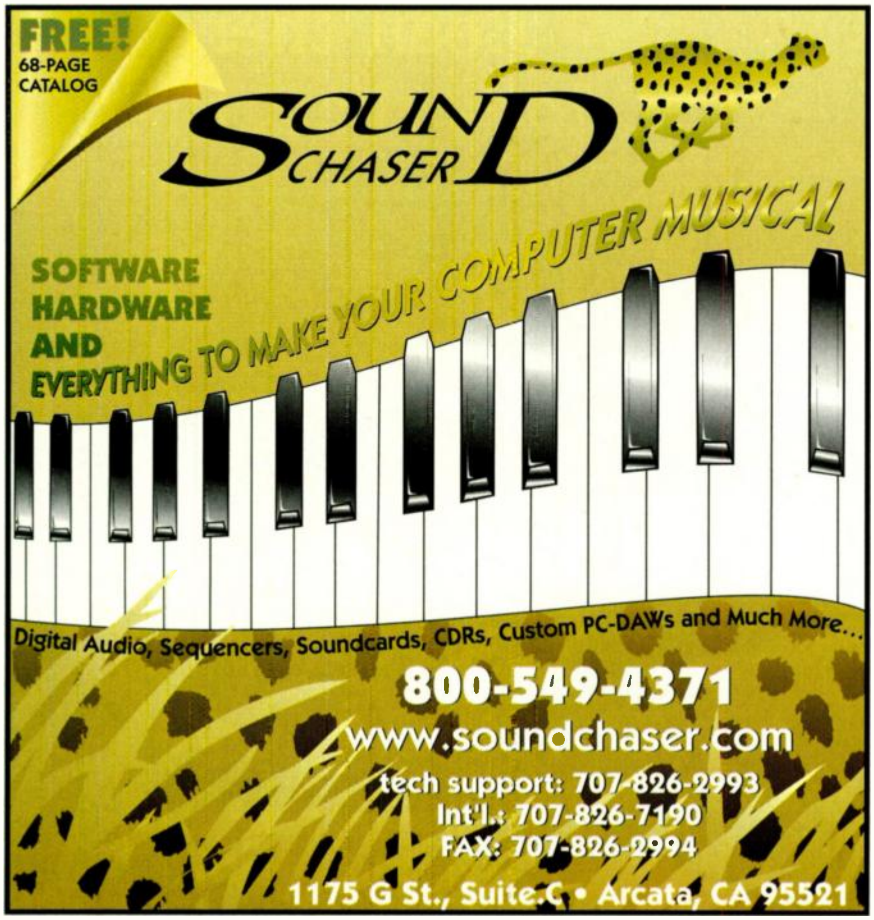

tests, the Shure KSM32 displayed a remarkably flat low-end response and superior bass extension and clarity. It also displayed a prominent midrange, which can cause it to sound honky or murky on some musical selections. The mic was quiet, living up to its respectable 13 dB self-noise figure.

The KSM32's superior resolution of room character, reverb, and other nuances was instantly noticeable in the mixes that I auditioned. It provided ambient details and a real sense of space in the mix, even when I used a relatively inexpensive boom box as the music source.

The mic's balance of accurate lows, rich mids, and pleasing, natural high-end response is very appealing, not to mention rare in a condenser microphone selling for around $1,000. Add the sonic benefits of transformerless, Class A electronics and the bonus incentives of a quality flight case and suspension shock-mount—not to mention the visual allure of this mic's sleek, futuristic styling—and you have a serious, end-of-the-millennium microphone bargain.

The KSM32 doesn't offer an equalized or souped-up sound, and for this reason it may not bowl you over the first time you hear it. The mic's naturalness could be one reason that it didn't produce great results for the vocalists I tested it on. But the KSM32's realism also sets it apart from the growing ranks of comparably priced cardioid condensers, a class of microphones that tends to offer personal-studio owners "enhanced" sonics (that is, puffed-up bass, glittering highs, scooped mids, and so on) rather than accurate, full-frequency reproduction.

I really enjoyed using this mic and greatly appreciated its true-to-life sound and Shure Brothers' top-quality workmanship. The KSM32 is a serious recording tool, and it is gratifying to see a major player such as Shure raising the industry standard by emphasizing sonics, features, and durability. Clearly, the company went all out to make a better, rather than simply cheaper, studio microphone.

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

ALESIS

DM PRO

A powerful and affordable drum module for pros.

By Brad Schlueter

Alesis has recently introduced the successor to its popular D4 and DM5 drum modules, the DM Pro. Surpassing its predecessors by leaps and bounds, the DM Pro is a thoroughly professional unit with features that read like a drummer's wish list.

The DM Pro houses 1,664 sounds, offers 64-note polyphony, and is 16-part multitimbral. It has a trigger-to-MIDI interface, six ¼-inch TRS outputs, dual onboard effects with 24-bit DACs (based on the Alesis Q20 engine), and positional hi-hat control. Using Alesis's *SoundBridge* software (which comes bundled with the unit), you can create your own sounds and sequences on your computer and import them into the DM Pro.

WHAT'S UP FRONT

The front panel of the DM Pro resembles that of the DM5. Housed in a single-unit rack-mount case, the module has a large backlit LCD screen, a value wheel, a ¼-inch headphone jack, a master volume knob, and buttons dedicated to various edit functions. These front-panel buttons are brightly backlit, making them easy to see on stage or in a darkened studio. The headphone amp is powerful, making it a blessing for drummers. Also located on the front panel is a PC Card expansion slot.

The DM Pro features a Preview button that allows you to audition sounds without using a MIDI controller or trigger pads. Unfortunately, the button is not Velocity sensitive (that is, it plays the sample at the maximum MIDI Volume of 127), so it doesn't let you audition the different Velocity levels of a given sound.

AROUND THE BACK

The back panel has six ¼-inch TRS audio outputs (two mains, two stereo auxes, and two mono auxes) and MIDI In, Out, and Thru ports (see **Fig. 1**). The DM Pro provides ten trigger inputs, six of which are TRS jacks for dual-zone pads (snare/rim, cymbal, ride, and hi-hat). The kick and three tom inputs are monaural. Although the DM Pro defaults to a three-tom kit, you can assign more toms if you wish, and you can also split a TRS input into dual mono inputs.

Two auxiliary RCA inputs allow you to monitor an external audio signal through the headphone or main outputs. However, the DM Pro has no volume control for the RCA inputs, so the volume must be set at the audio device itself. The unit is powered by an uncommon 4-pin DIN wall wart.

ENVELOPES PLEASE

The DM Pro provides two distinct editing modes: Drumkit, in which you create kits and tweak a limited number of common parameters, and Drum Edit, which allows editing of higher complexity.

In Drumkit mode you choose which instruments you want in your kit, adjust their volume, tune them, pan them across seven positions, and select an output to send each instrument through. You can set two drums to sound simultaneously whenever a single pad is struck or a MIDI note message is received. Or you can assign one sound to cut off another, to achieve effects such as open/muted triangles or choked cymbals.

Detailed editing takes place in Drum Edit mode. Individual sounds are called Drums, whether they are actual drum samples or not. Each Drum is made up of four Sounds, each of which has Filter, Envelope, and Modulation settings. There are three Envelopes per Sound: Pitch, Filter, and Amplitude. Using the unit's Modulation Matrix, you can have any of these Envelopes control or modulate a number of other parameters. All told, 16 modulation sources control 21 destination parameters.

In addition, the DM Pro has four Trigger Parameters per Envelope that can drastically alter the Envelope's behavior. Some of the parameters give you synthesizer-like control over the sounds, while others offer more subtle musical control. For example, the Pitch Envelope is useful for changing the pitch of talking drums, tablas, and conventional toms.

BIG EFFECTS

The DM Pro has two effects buses—Reverb and Multi-Effects—that can be run in serial or in parallel. Reverb offers gated, plate, room, large, hall, and reverse reverbs; the multi-effects processor provides delay, overdrive, and pitch effects (chorus, flanger, and resonator). The output of the multi-effects processor can be sent through the reverb processor, although it doesn't work the other way around.

All of the effects in the DM Pro are 24-bit resolution, and they sound great. The reverb has ten editable parameters, including Density, Decay Time, and High-Frequency Decay (which darkens the reverb's timbre). The delays in the unit are monaural, so there are no ping-pong effects. Editable delay parameters include Output Level, Input Mix, Delay Time (adjustable in 1 microsecond increments), and Feedback. Finally, it has a 2-band boost-only shelving EQ that globally affects the entire Drumkit.

Each Drumkit has 64 MIDI notes assigned to it. Each of these MIDI notes has an effects-bus and effects-send level assignment. If you change a Drum assigned to a given MIDI note, the new sound will have the same effect parameters as the previous one.

Drums sent through the main outputs can have effects assigned to them. Sounds sent through the stereo aux outputs are sent dry, while any effects assigned to them are sent through the

The Alesis DM Pro delivers a wealth of drum and percussion sounds and a number of features geared toward the professional percussionist.


WHAT'S ALL THIS ABOUT HAFLER?
There's a buzz on the street... Hafler's the one to use. And there's good reason.
Hafler has been building reference studio amplifiers for over 25 years. Innovative technology and forward thinking have always been the difference between Hafler and everything else. Patented technologies like trans•nova and Diamond that provide a simpler gain path for lower noise and distortion. Technologies that have made Hafler the music professional's choice in studio amplifiers for years. Technological advantages that no one else has.
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FIG. 1: The DM Pro has ten trigger inputs, six of which are optimized for dual-zone pads such as snare, cymbal, and ride, as well as a hi-hat controller pedal.

main outputs. Sounds going through the mono aux outputs cannot have effects assigned to them. For this reason, many of the drums have been sampled with an ambience.

BEAT THE DRUMS

Although the DM Pro's sounds are geared mainly toward dance and pop styles, Alesis has also supplied several excellent ethnic and orchestral samples. The sounds are organized into 13 categories called Drum Groups: Acoustic Kicks, Electric Kicks, Acoustic Snares, Electric Snares, Toms, Hi-Hats, Cymbals, Acoustic Percussion 1 and 2, Electronic Percussion, Special FX, Chromatic, and User. Each Drum Group contains 128 sounds.

The sounds in the Acoustic Snares group are generally good, although some of the samples are a little too "wet" for my taste. You can remedy this on some samples by lowering the reverb send; on others, though, the ambience is part of the sample and cannot be removed. The brush snares are nice, but no brush sweep is included. Many of the snare sounds have a filter that opens and brightens the sound at higher velocities, imitating the way real snare drums behave.

The Electric Snares group sounds good; it includes 808, 909, Simmons, Rap, Rave, and House snares, plus some sounds best described as zappy, ringy, metallic, gated, and explosive. The DM Pro's Acoustic Kicks and Electric Kick collections contain many useful bass drums—and then there are the subharmonic House and Rap sounds found in the User group.

Alesis combines electric and acoustic toms into one Drum Group, so although you get 128 sounds from each of the other sound categories, there are only 64 sounds of each type of tom, all lumped into one menu. This is unfortunate; I'd rather have one menu dedicated entirely to acoustic toms.

Only two of the tom sets sound realistic enough for general-purpose rock/pop studio work; the rest I would probably never use. For instance, the Jazz toms have more of a rock sound than I expected; the Brush toms have a synthetic attack; and the Pop, Floppy, Klasse, and Live toms all have two different pitches occurring simultaneously. There are two great Roto tom samples, but Alesis chose to include only small Rotos, which sound artificial when tuned low. The Electric toms include Simmons-style, 808, Synsonic, and a lot of weird sounds. The Blaster toms could easily work in a sci-fi soundtrack as phaser effects.

The cymbal sounds in the DM Pro are a huge improvement over those in previous Alesis drum modules. The Cymbals group includes a nice variety of timbres: cymbals with rivets, chinas, splashes, gongs, flanged cymbals, reversed cymbals, choked crashes, orchestral crashes, and mallet rolls. Unlike with the D4 and DM5 units, the DM Pro's cymbal decays are not too short, and the loops are generally imperceptible. And yes, the hi-hats respond realistically to hi-hat controllers such as Roland's FD-7.

The 256 Acoustic Percussion sounds are original and excellent. This group includes tablas, taiko, and udu drums; shakers; flexatone; finger cymbals; castanet rolls; wind chimes and bells; and all the Latin sounds you'll ever need. There's even a sample of two drumsticks being struck together for song count-offs.

Most of the percussion samples are Velocity sensitive. For example, the hand drums gain brightness and attack as you play harder; other instruments, including Back Beat and Ringer, have sharper attacks; and still others, such as VeloConga and Skinner, change pitch subtly. Certain samples are of acoustic timbres, but they are processed in such a way that they sound very electronic. The dedicated Electronic Percussion group contains a wide variety of sounds ranging from 808-style congas, claves, and hand claps to a metronome click and a submarine sonar blip.

Special FX includes everyday sounds such as door chimes, squeeze toys, and anvils, as well as some esoteric ones. Some of the samples are short performances using acoustic timbres; for example, Horsey is the sound of coconuts being banged together to simulate a horse's gallop. The Special FX group really shows off the power of the DM Pro's envelope filters: some sounds evolve slowly over time, while others sustain almost indefinitely.

Theater percussionists should note that the unit's pitched percussion sounds are very good. And unlike earlier Alesis drum modules, the DM Pro comes with a generous selection of orchestral instruments, including timpani, vibraphone, xylophone, and marimba. A collection of other pitched-instrument sounds, such as guitar, bass, and synth, is also included.

The organization of sounds within some groups could be better. Certain sounds are thoughtfully subgrouped together—including, for instance, all

DM Pro Specifications

Synthesis Engine	sample playback
Maximum Polyphony	64 notes
Number of Drums	64/16
Multitimbral Parts	16
Patches (ROM/RAM)	1,536/128
Effects Types	reverb; overdrive; delay; pitch; EQ
Audio Outputs	(6) ¼" TRS; (1) stereo ¼" headphone
Audio Inputs	(2) RCA
Trigger Inputs	(6) ¼" TRS; (4) ¼" unbalanced
Other Ports	MIDI In, Out, Thru; (1) ¼" aux pedal; (1) ¼" Noise Suppression; (1) 4-pin DIN power plug
Expansion Slots	(1) PC Card expansion slot
Dimensions	1U x 6" (D)
Weight	4 lbs., 8 oz.

For manufacturer information, please see Contact Sheet, p.179.

variations of the Chromatic menu (Timpani, Vibraphone, and so on). However, the sounds are usually not alphabetized. With more than 1,600 samples available, this makes finding an individual sound difficult.

MIDI TO THE MAX

The DM Pro has extensive MIDI features that are flexible enough for any project. They are divided into two areas: Drumkit MIDI functions and Global MIDI functions.

The Drumkit MIDI functions include Drum Channel Override, MIDI Input Enable, and MIDI Output Enable parameters. In the default setting, a Drumkit responds to a single MIDI channel. However, by using Drum Channel Override, the DM Pro can transmit or receive data on 16 MIDI channels at once, allowing you to send specific MIDI controller commands to specific instruments. For example, you can send pitch bend information to the toms only, change the panning of the shakers, or alter an instrument's volume from a sequencer or a continuous controller. You can enable or disable instruments' receiving and transmitting of MIDI data within a Drumkit, as well.

The DM Pro's Global MIDI menu offers 11 editable parameters. Filter Program Transmit controls whether Program Change messages are sent from the MIDI output. Similarly, MIDI Start On/Off toggles the unit's ability to send sequence start commands from the triggers. Hat Pedal Controller Assign allows an external controller's Modulation Wheel to control the hi-hat pedal, while Controller A-D Assign lets you assign four separate Continuous Controllers to alter parameters within the DM Pro's Modulation Matrix.

WHOA, TRIGGER

The DM Pro has 12 parameters that affect triggering. This is a fairly flexible arrangement; for instance, you can assign independent triggers to start and stop an external sequencer, or assign a single trigger to handle both tasks. (Doing the latter frees up a trigger input.)

In addition, there are Noise Suppression and Crosstalk parameters in the unit's Trigger menu. Noise Suppression prevents false triggering caused by vibrations. This feature cleverly defines the noise floor threshold of the

ALESIS
DM Pro drum module
$899

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

PROS: Excellent sounds. Four customizable trigger setups. Powerful MIDI implementation. Great Trigger-to-MIDI interface. Extensive modulation control. PC Card expansion slot.
CONS: Takes about a second to load kits. Loading requires an extra button push. Challenging to edit. Has a 4-pin DIN wall wart.
CIRCLE #440 ON READER SERVICE CARD

environment so that the DM Pro will trigger only when a signal exceeds that threshold. Crosstalk automatically adjusts the triggering threshold in a kit, keeping unintended triggers from firing when another is struck. Only one of these features is available at a time, but both work well. I put a trigger on an acoustic tom to see if I could detect any triggering anomalies; it was right on the money and didn't need tweaking. Triggering was fast, and I could detect no delay between striking and sound reproduction.

Triggering from tape is a great way to "fix it in the mix." Thirteen velocity curves allow you to customize the unit's response to best suit the track material and the drummer's playing style. By adjusting the DM Pro's sensitivity, you can also remove unintentional ghost notes from a taped performance just as a noise gate would. In addition, the DM Pro offers a flat, maximum velocity curve for dance tracks that don't need dynamics.

Alesis has included positional hi-hat control in the DM Pro. When used with a continuous controller pedal, the unit can accurately emulate an acoustic hi-hat, with open-to-close, foot closure, and splashed hi-hat variations.

My favorite triggering feature is Trigger Setup Select, which enables you to save up to four custom trigger setups. If you use your DM Pro with different pad combinations, it can store each of these unique trigger profiles. And kudos to Alesis for designing the DM Pro so that you don't lose user-defined banks of Drumkits, Drums, or custom trigger setups when you reinitialize the unit.

BUNDLES OF JOY

The DM Pro comes with a CD-ROM of samples, sequences, and cross-platform applications. One of them is *SoundBridge,* which allows you to transfer samples and sequences from your computer to the drum module. Using MIDI SysEx data, files are sent to the DM Pro and saved to the PC Card. Because the DM Pro has no sequencing or sampling abilities of its own, this application is especially useful. (For a Macintosh, *SoundBridge* requires a 68030 processor running Mac OS 7.1 or higher; on the PC side, you need Windows 95, 98, or NT4. At least 3 MB of RAM is recommended.)

The PC Card can hold up to 8 MB of MIDI Song Files, as well as AIFF, WAV, and other sound files (depending on the size of the card). Triggers can be assigned to start or stop a sequence stored on a PC Card; however, MIDI data does not transmit when sequences are played from the card.

CLEARLY A PRO

Alesis clearly has a winner with the DM Pro. This is a powerful unit for the money—at twice the price of the DM5, you get three times the number of sounds (which, except for a few clinkers, are excellent). You also get killer effects and a host of innovative features. The user manual is very thorough and contains a helpful index. The inclusion of *SoundBridge* greatly increases the power of the unit; if you like the nitty-gritty of sound design, you'll find enough editing potential to keep your sounds fresh for years to come. The DM Pro is a wise choice for any number of professional applications.

When **Brad Schlueter** *isn't gigging or teaching, he wears a kilt and practices the difficult art of Scottish snare drumming.*

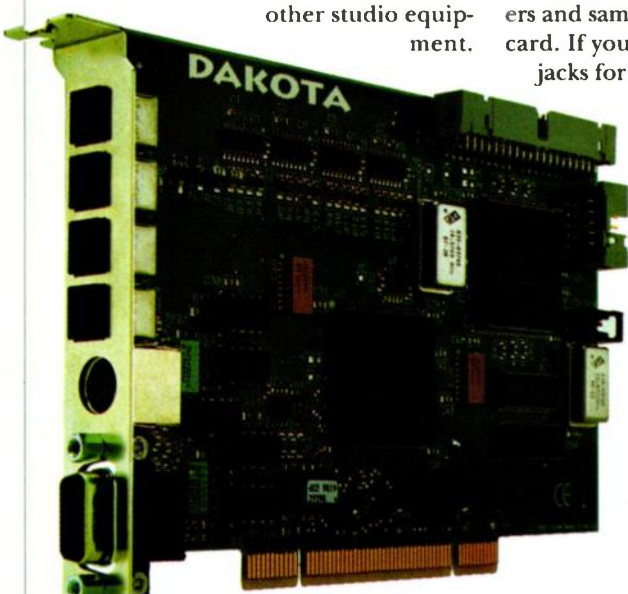
FRONTIER DESIGN

DAKOTA (WIN) AND TANGO24

A versatile, expandable digital audio system with something for everyone.

By Zack Price

Let me say this right up front: I prefer using multichannel digital-only audio cards in my work, and that typically means (almost by default) a card with 8-channel Lightpipe optical I/O. I realize that this option can be more expensive than using prepackaged multichannel cards with dedicated breakout boxes, but digital-only audio cards offer some distinct advantages. First, they can interface with a wide variety of compatible outboard components, such as MDMs, digital mixers, multichannel A/D/A converters, synthesizers, and samplers. Second, you can use long, easily replaceable optical cables to run between the card and the external devices. This lets you use cables that are optically isolated, thereby avoiding ground loops, and the comparatively greater cable lengths simplify the placement of outboard gear in relation to the computer and other studio equipment.



Frontier Design Group's versatile Dakota offers digital audio with ADAT Optical, S/PDIF, and MIDI I/O, as well as a variety of sync options.

Currently, most manufacturers of multichannel digital-only cards offer products that are limited to providing Lightpipe connections for transferring digital audio between the computer and outboard devices. However, Frontier Design Group has taken that design concept to the next level with its powerful new Dakota card. It provides 18-channel digital audio capability (16 ADAT and 2 S/PDIF), in addition to a variety of other MIDI and synchronization features. Moreover, Frontier also makes a line of optional equipment—including the versatile Tango24 multichannel A/D/A converter—that can make the Dakota the hub of a powerful desktop music production system.

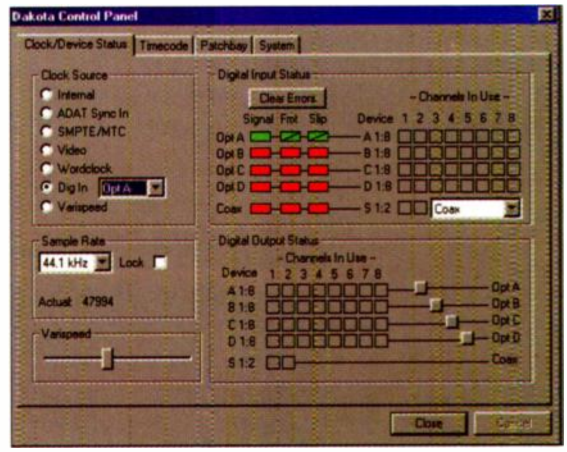


FIG. 1: The Dakota's control panel allows you to set and view Clock/Device Status, Timecode, Patchbay, and System parameters. It also shows when the card's input or output status is valid and alerts you to status errors, as indicated in this illustration.

THE SCENIC ROUTE

The Dakota comes in the form of a short PCI card configured with two optical inputs and two optical outputs for its 16 ADAT-compatible digital audio channels. Either pair of optical jacks can be switched to optical S/PDIF from within the Dakota's software control panel (see **Fig. 1**). This allows you to connect DAT decks and CD and MiniDisc players, as well as some synthesizers and samplers, directly to the Dakota card. If you need both pairs of optical jacks for 16-channel operation (or if you just need more channels to work with), you can add Frontier's Montana daughtercard to your system (see the sidebar "Montana Wildhack").

A 15-pin connector on the Dakota's faceplate connects to a breakout cable for electrical S/PDIF (coaxial) I/O, which is switchable to AES/EBU from within the control panel. (You'll also need to use properly wired cables between any AES/EBU device and the RCA jacks of the Dakota card.) The breakout cable also includes a 9-pin ADAT-sync input jack. This turns your computer into the last slave device in an ADAT chain so that you can run your digital audio software in sync with your ADAT-compatible MDMs from the master device in the chain.

An 8-pin connector for MIDI I/O is also provided on the faceplate. You can connect the included breakout cable with the two pairs of MIDI Ins and Outs for a total of 32 MIDI channels. Alternatively, you can connect Frontier's Sierra 8 × 8 MIDI interface/patch bay to this jack for 128 MIDI channels (see the sidebar "High Sierra").

The Dakota also contains two internal connectors. The first is a digital-audio input that connects to the digital outputs of compatible CD-ROM drives for digitally transferring sound in real time to appropriate audio programs. The second connector is an internal 40-pin header that links the Dakota to the optional Montana daughtercard for expanding its audio and sync capabilities.

TWO TO TANGO

Because the Dakota is a digital-only card, you must employ outboard devices to record and play analog audio through the card. If you plan only to transfer tracks between your ADATs and the Dakota, then your setup is complete with just the card. However, you can't use an ADAT as a front-end device if you need to overdub individual tracks, because a single ADAT can't be used for both A/D and D/A at the same time; the analog/digital input selection affects all eight channels at

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FACE THE FACE

The Tango24's front panel provides a variety of toggle switches and informative status lights. Two switches on the left let you select the sampling rate (44.1 or 48 kHz) and the clock source (word-clock input, ADAT Optical input, or internal). Next to the Clock Source switch, the Clock Status indicator light glows steadily when the clock source is stable and blinks when the external clock source is outside the Tango24's frequency-locking range. The light also blinks when there are other clocking errors, such as an incorrect format.

The Optical Input indicator shows the status of the ADAT Optical input signal. It glows steadily when the input is valid and locked to the clock system. When the light blinks slowly, it means that the optical input is valid but is not locked to the Tango24 clock system. Most often, this occurs because the Tango24's optical input is based on a different clock source and is "slipping." (For example, the Tango24 may be receiving digital audio data at 48 kHz when its own sample frequency is set to 44.1 kHz.) You can confirm this by viewing the Clock/Device Status tab in the Dakota's control panel. **Figure 1** shows how the tab appears when the digital audio data comes in at a rate that is different from the indicated settings.

When the Optical Input light blinks quickly, the data stream itself has errors. This usually happens when the Tango24 is not receiving ADAT Optical format data—because, for example, you accidentally routed two-channel S/PDIF digital audio through the optical ports of the Dakota or Montana card to the Tango24.

Three rows of eight LEDs serve as

Tango24 Specifications

Resolution	24 bit
Sample Rates	39–51 kHz from optical or word-clock input
Frequency Response	20 Hz–20 kHz, ±0.05 dB
THD+N (EIAJ)	0.002%, A weighted
Signal-to-Noise Ratio (EIAJ)	105 dB
Analog Connectors	balanced ¼-inch TRS phone jacks
Reference Level	+4 dBu or -10 dBV selectable per channel
Headroom	15 dB (maximum 5 dBV at -10 dBV reference level or 19 dBu at +4 dBu reference level)
Clock Sources	internal (44.1 or 48 kHz), word clock, optical input
ADAT Optical Ports	in, out, thru
Word-Clock Ports	in, out
Power Supply	10–12 VAC with barrel jack connector
Dimensions	1U x 8.5" (D)
Weight	7 lbs., 8 oz., including AC adapter

either input or output level meters, depending on how you set the neighboring Meters switch. The lights in the bottom row, labeled Signal, glow green whenever a signal is present. The intensity of the lights varies in proportion to the signal strength. The middle row of yellow lights indicates when the signal is within 3 dB of full-scale digital signal (0 dB). These lights remain lit for about half a second unless the signal stays constantly within 3 dB. The red LEDs in the top row, labeled Clip, light up when the signal is within 0.002 dB of full scale. When metering outputs, these LEDs remain lit for about half a second. When metering inputs, however, the clip indicators stay lit until you reset them by toggling the Meters switch to Output and then back to Input.

The Tango24's power is supplied by a hefty lump-in-the-line AC adapter. There's no on/off switch, but you can tell that the Tango24 is powered up when the green power indicator on the left is lit.

The Tango24 looks pretty cool when all its LEDs are glowing in the dark, but how does it sound? In a word, great! Though I performed no bench tests to confirm or disprove the unit's specs, I did record and play back a variety of audio material that I am familiar with. The Tango24's balanced analog and optical digital I/O made for very clean recordings. In fact, I felt comfortable enough with its quality that I used it in recording sessions for a client who uses both acoustic and electronic instruments in his work.

CONTROL FREAK

Version 2.0 of the Dakota control panel installs drivers for Windows 95 and 98, along with ASIO 2.0 drivers for use with Steinberg's *Cubase VST* and other ASIO-enabled applications. The Dakota 2.0 driver supports sample-accurate I/O using Syntrillium *Cool Edit Pro* 1.2 (the *SE* version of which is included in the Dakota package). Unfortunately, there are no DirectX drivers for the Dakota at this time, although a Frontier spokesman stated that the company is actively developing them. I hope it does, because many software synthesizers and other products—most notably NemeSys *GigaSampler*—use DirectX.

The Dakota control panel itself is now accessible from the Windows 95/98 Taskbar. When you right-click on its icon, you see a list of the tabs in the control panel (Clock/Device Status, Timecode, Patchbay, and System). When you click on a selected tab name, the control panel opens with that tab showing. It may not seem like much, but it's a handy time-saver.

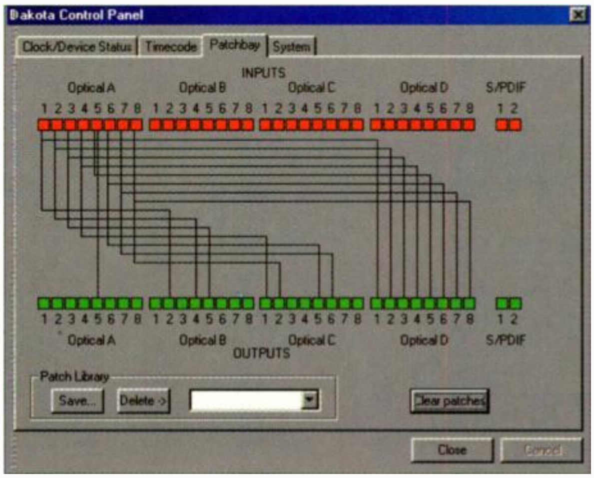

FIG. 2: The Patchbay tab in the Dakota's control panel lets you route any input from the Dakota or Montana to any output of either card. The settings can also be saved for later recall.

The System tab displays the current Dakota software version as well as information on contacting Frontier Design Group. It also has a MIDI mode section for matching the MIDI I/O to the device being used: 2 × 2 when using the supplied MIDI breakout cable, or 8 × 8 when connected to the Sierra MIDI interface.

The Patchbay tab lets you route any input to any output of the Dakota or the Montana daughtercard (see **Fig. 2**). Furthermore, you can route an input to more than one output. For example, I have routed the inputs of Optical Port A directly to its outputs. I have also routed each channel in Optical Port A to selected individual channels of the other three optical ports. This lets me monitor the eight incoming channels and send a copy of each channel to other devices in the system. You could consider this feature a kind of simplified software version of an Alesis BRC. Best of all, you can save each patch-bay setup for easy recall.

The Clock/Device Status tab is where you select the clock source for the Dakota. If you're using the Dakota as the master clock, you set the clock source to Internal. The Dakota also receives clock information from a variety of sources, depending on the types of devices you have in your system. It can accept ADAT sync input from an external ADAT, for example, and it also accepts SMPTE/MTC if you have the Sierra MIDI interface connected to the Dakota.

With the Montana daughtercard installed, the Dakota can receive video sync or word clock. When the Digital In source is selected, the Dakota derives its clock source through whichever optical or coaxial inputs you select. For instance, you can use the Tango24's clock through its optical jack, or use the digital output of a DAT deck connected to the coaxial input on the Dakota. When Varispeed is selected, you can adjust the sample rate with the Varispeed slider. This option is useful if you need to pull down the

Dakota Specifications

Digital Audio Inputs	16 channels ADAT Optical; 2 channels S/PDIF (either coaxial RCA or Toslink optical)
Digital Audio Outputs	16 channels ADAT Optical; 2 channels S/PDIF on coaxial RCA and/or Toslink optical
Resolution	8, 16, 20, and 24 bit
Sample Rates	44.1 and 48 kHz internal; Varispeed and input tracking from 39–51 kHz; real-time resampling to support 8, 11.025, 16, 22.05, and 32 kHz output streams
MIDI I/O	(2) MIDI In, (2) MIDI Out
ADAT Sync Input	DB-9 connector via breakout cable from HD-15 port
SMPTE Time Code	any audio input or output can be used for SMPTE time code

sample rate to match the frame rate of some types of video decks. (For more on pulldown, check out "Square One: Picture Perfect Sound" on p. 96.)

The Dakota control panel's Digital Input Status section also displays the audio input status, mappings from physical inputs to logical inputs, and channel indicators. Each optical input of the Dakota and Montana (along with the coaxial input) has three indicator "lights." (The indicators for the Montana are always present, even when the card is not installed.) The left light indicates if there is an active input signal, and the middle light shows whether a valid digital audio format is present.

The right light is the Slip indicator. It appears green when the Dakota is locked to an incoming audio clock. If it turns red, it means that the Dakota's sample rate is different from the incoming audio clock. If the Dakota's rate is almost the same but not perfectly locked, the Slip indicator may flash red periodically. A slash will also appear across the green indicator until you click the Clear Errors button.

The Actual reading in the Sample Rate section can show you whether the current clock source is inactive or unstable. For example, if you've selected Digital In–S/PDIF as the Dakota clock source but your DAT machine is also set to digital input, then neither device is acting as a master (each is trying to lock to digital input from the other); in this case, the Dakota clock system will spin down to its lowest frequency. Switching the DAT back to analog input solves the problem, and you'll see the actual sample rate settle in to the proper 44.1 or 48 kHz.

By default, the Dakota's physical inputs are mapped directly to logical devices. For instance, Optical Port A maps to Logical Device channels A1 through A8, Optical Port B maps to Logical Device channels B1 through B8, and Coax maps to S1:2 (S/PDIF). Device S1:2, however, can come from any physical input. Using the drop-down box next to the device, you can choose any of the four optical inputs, the coaxial (RCA) input, or the CD-ROM; the physical-to-logical mapping lines automatically change to reflect the input status.

You can use the Tango24 to send digital audio twice as far as the normal limit.

The Digital Output Status indicator simply shows which output channels are active along with the mapping from logical output devices to physical outputs. Although the stereo S/PDIF stream is always present on the Dakota's coaxial output, it is also possible to route S/PDIF to any of the four optical outputs: just click on the button next to the corresponding optical output, and it changes from 8-channel to 2-channel mode. Similarly, the logical-to-physical mapped lines automatically change to reflect the output status.

TIME IS ON MY SIDE

The Timecode tab is one of the most important features of the Dakota card. It offers a choice of several time-code sources, so there is bound to be a method of synchronization that you can use. Typically, the time-code source is set to Internal so that the PC's applications can control time-code generation. If ADAT Sync In is chosen, the Dakota automatically converts to MIDI Time Code. That's because most recording/playback software can receive and follow only MIDI Time Code. (It's the only form of time code in the Windows standard interface.) To receive MTC from a MIDI input, select the MTC Port option and the appropriate port number (1 or 2 if you are using the Dakota's two-In/two-Out MIDI cable).

By far the most interesting Timecode Source selection is the SoDA Channel option. SoDA stands for "SMPTE on Digital Audio," and with this option you can receive SMPTE time code through any Dakota (or Montana) digital audio input. Just connect a SMPTE time-code source to an A/D converter (such as the Tango24) and then to the desired Dakota or Montana digital audio input.

Dakota
Minimum System Requirements
Pentium-class CPU; Windows 95/98; PCI card slot

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Tango24 8-channel A/D/A converter
$699
Dakota and Tango24 package
$1,149

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

PROS: Excellent sound quality. Built-in MIDI support. Flexible time-code and sync options. Devices interconnect with a wide variety of third-party products. Expandable with other Frontier products.
CONS: No DirectX driver. No 96 kHz recording option.
CIRCLE #441 ON READER SERVICE CARD

Next, set Timecode Source to SoDA Channel, and then choose the appropriate input channel from the adjacent selection box.

To use SoDA for sending time code from a digital audio output, you simply select the SoDA Channel box from the Timecode Destinations section; then choose the desired output channel and set the output level of the SoDA channel signal.

If you use the SoDA Channel input and output options to synchronize the Dakota to a multitrack analog tape deck, you'll probably need to stripe the tape first. That's easy enough with the Dakota's Manual Stripe to SMPTE/MTC feature.

SUM OF THE PARTS

A digital-only card's value is largely determined by the amount and variety of outboard equipment that can be connected to it. By itself, the Dakota matches the competition based on its digital audio features alone. What puts the Dakota ahead of most of its rivals is its stand-alone MIDI capability. Moreover, very few competitors offer ADAT synchronization features beyond the ability to sync the sample rate of the card to a Lightpipe input.

The Dakota really comes into its own, however, when used with Frontier's other products. As I mentioned earlier, the Montana daughtercard adds 16 more ADAT channels, as well as video/word-clock input synchronization and ADAT sync output, so the computer can be placed in any slave position in an ADAT chain. The Sierra MIDI interface expands the Dakota's MIDI capabilities even further and provides SMPTE in and out. And the Tango24 converter gives you eight channels of A/D/A, along with word-clock in and out, at a very reasonable price. What's more, it sounds great.

The only real drawback to using a customizable system like this is that it initially costs more than a prepackaged audio card/multichannel box. Even so, the price of the Dakota/Tango24 bundle competes well with those of prepackaged setups. And when you consider that the Dakota/Tango24 combo offers more flexibility and expandability, the price difference pales in comparison to the difference in value. Besides, the Montana and Sierra hardware options are relatively inexpensive, and you can buy them as your needs and budget dictate.

The only specific black marks that I can give to the Dakota are its current lack of DirectX drivers (although Frontier hopes to have one by the time you read this) and its inability to sample at 96 kHz. However, most other multichannel cards don't sample at that rate, either. Besides, given the current level of technology, I don't want to sacrifice audio tracks or overburden my computer just to move up to a standard that is often used mainly for marketing hype.

Beyond these gripes, I have nothing bad to say about the Dakota, Montana, and Tango24 devices. In fact, I am so impressed with this system that it is the next planned purchase for my studio.

Zack Price *is a digital audio editor and Windows digital audio consultant in the Chicago area.*

MONTANA WILDHACK

To those who never seem to have enough, Frontier Design Group offers the Montana daughtercard ($249), which adds 16 more channels of digital audio to the Dakota card. The Montana has two pairs of optical I/O, and as with the Dakota, each can be used for 8-channel Lightpipe or 2-channel S/PDIF digital audio. In addition, the Montana has a 9-pin connector for ADAT sync output. When both the Montana and the Dakota are installed, the computer can function as a slave device anywhere within an ADAT chain. This allows the computer to follow incoming ADAT time code without affecting the operation of other ADAT devices down the chain.

The Montana also includes an RCA connector that can be used for video sync or word-clock input. Although any audio RCA cable can connect to the video/word-clock input, you should use a 75-ohm coaxial video cable. Alternatively, you can use a BNC cable with the Montana's supplied BNC/RCA adapter. The Montana also comes with an internal video connector that can attach to digital video cards like Pinnacle Systems' miroVideo DC-30. Keep in mind, though, that you can't use the internal and external video connectors at the same time.

Because of its unique design, the Montana can be installed in either a PCI or ISA slot adjacent to the Dakota card. The Montana comes from the factory ready to be put into a PCI slot; however, if you turn the card upside-down and exchange its faceplate with the supplied alternate, you can install it into an ISA slot. In both instances, the Montana fits into the 40-pin connector on the Dakota card. (The connecting cable is extremely short, which is why you have to install the Montana in a card slot adjacent to the Dakota.) Once the Montana is installed, the Dakota's drivers and its software control panel are able to automatically recognize it. Furthermore, no additional IRQs are required.

JOEMEEK

C2

Proof that good things do come in small packages.

By John Ferenzik

Photoelectric—or photo-optical—compressors have always held a certain mystique. They've worked aural magic on some of the most legendary tracks. The best of the vintage units, as well as their modern equivalents, are renowned for their transparent sound, low noise, low overall distortion (even when compressing at extreme levels), and a certain musical unpredictability when handling transient peaks.

Having heard such vintage wizardry applied in the studio, I was eager to try the new Joemeek C2 photo-optical stereo compressor. The C2 is a half-rackspace unit that employs the same photo-electric sensing device that is used in its bigger siblings, the SC2 and SC2.2, but with several new design twists. At first glance, the unit's small size and the simplicity of its front-panel controls suggest "personal studio" rather than "fully professional." Looks, however, can be deceiving: this little emerald glistens on many applications. And it *is* small, so you can easily transport it from studio to gig and back without fuss.

PHOTO FINISH

With its green front panel and solitary red button, the C2 bears the unmistakable marks of its Joemeek lineage. It has two LED meters—one for gain reduction and one for input levels—and five knobs. From left to right, the knobs control input gain, compression (in *Spinal Tap* tradition, it goes all the way to 11), attack time (from 1 to 11 ms), release time (from 250 ms to 3 seconds), and output volume. The red button toggles the compression effect on and off. (There is no on/off switch—just a 12 VAC wall-wart power supply for plug-and-play simplicity.)

So, you might be asking, where is the ratio control? Well, behind the C2's Spartan facade beats an interactive heart. One of the design innovations on this product is that the compression ratio (which ranges from 2:1 to 14:1) is controlled by the input gain. There's no way to know the exact ratio; instead, you just have to experiment with the relationship between the Input and Compression knobs, listen to the results, and find what works best for the source material. Adjusting the amount of compression, attack and release times, and output volume has an audible effect, of course, but I found that altering the relationship between the Input and Compression knobs produced the most dramatic results in getting the C2 to put its photoelectric spin on my tracks.

The C2 also differs from other Joemeek compressors in that, at extreme input-gain levels, the unit acts as a limiter (typically, this happens as the ratio exceeds 10:1). However, as a limiter it exhibits a soft touch on transients: peaks still sometimes get their foot in the C2's door, even at the most extreme settings.

The C2's rear panel is about as simple as they come. It provides balanced, line-level stereo inputs and outputs on ¼-inch TRS jacks, as well as a separate jack for the power adapter.

The compressor's user manual made for an undaunting (even entertaining) read, grammatical anomalies notwithstanding. The guide is very informative for the personal-studio novice, while managing to include some helpful suggestions for the more experienced hand, as well.

GUITARIFFIC!

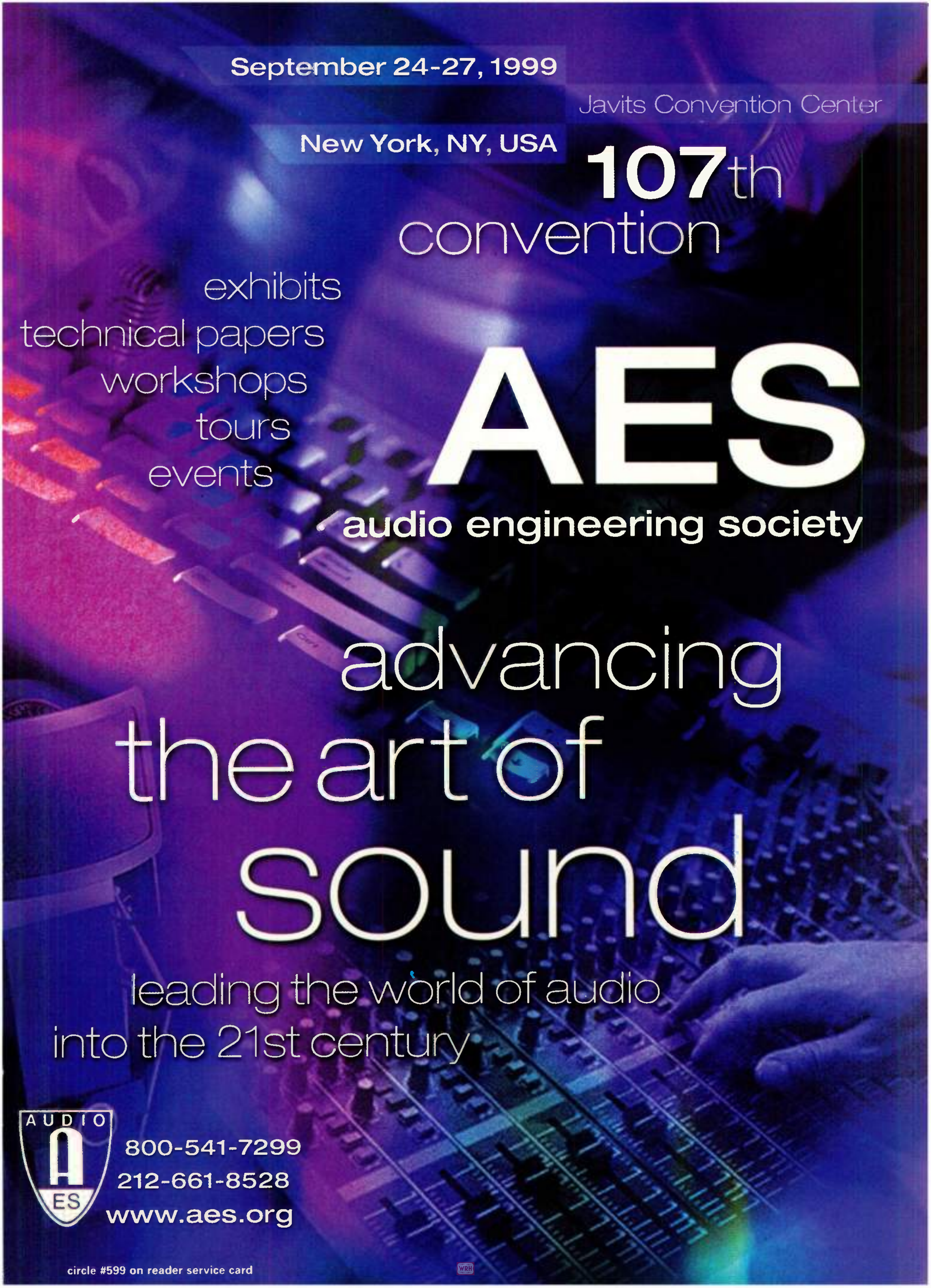
The C2 really sparkled on electric guitar tracks, both during recording and at mixdown. Initially, I used it on a session for which I had to overdub a series of electric guitars playing twangy, country-western-style licks. The C2 did a wonderful job of warming up the overdubs and evening out the bottom end without the characteristic flatness that comes from using VCA compression. I also liked the stomp-box flavor that it imparted to the guitar—only without the stomp-box noise. The C2 was very quiet in this application.

Next, with the C2 patched into the insert on a mixer channel, I recorded several passes of a loud, raspy, distorted guitar solo using an amp cranked up in another room. Listening to the playback, I was impressed by how the C2 let the spikes of the pick attack come through, even when I had shortened the attack time to the lowest setting of 1 millisecond (the unit has no zero setting). Although the transients didn't slip by totally unscathed—I noticed a subtle damping—the tracks retained the aggressiveness of the original performance. Meanwhile, the C2 brought up the rear, so to speak: the lower and middle frequencies leveled out behind the frontal pick attack. This served the guitar solo well.

In mixdown applications, I found the C2 handy both for heating up individual tracks and for compressing doubled parts in stereo. The unit is a bit cantankerous, though, when used as two separate mono compressors. Though this is something the manual expressly says *not* to attempt, I tried it anyway, processing two different-style guitar performances side by side. (Specifically, one guitar part played on the upbeats, and a second doubled the part but played with added percussive "chunks" on the downbeats.) The solution in this case lay in backing off the C2's input and shortening the release time. When applied in this manner (despite the user guide's warning), the diminutive box was able to wed the two disparate musical performances seamlessly.

Again ignoring the manual's stipulation, I tested the C2 on acoustic guitar. I laid down a strummed steel-string rhythm part to one track, overdubbed a solo line on another, and

The Joemeek C2 photo-optical stereo compressor is effective on a surprisingly wide range of sources, but it sometimes requires considerable tweaking to get just the right setting. Its sound ranges from smooth and transparent to aggressive and edgy.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then applied the stereo C2 on playback to the two separate performances simultaneously. Again, it took a lot of time and effort to obtain usable results for the two disparate tracks without squashing the music, but I was able to pull it off quite satisfactorily using the C2.

LOWDOWN ON THE BASS

Bass guitar can be a tricky animal to tame: too little compression and the track will exhibit inconsistent levels, peaking out from behind the bass drum occasionally and unpredictably; too much compression and the resultant sound is flat and unassertive. I was already in the middle of remixing some electric bass tracks, so I used the opportunity to audition the C2 in this difficult application.

I found out something interesting: the amount of discernible compression on the bass was directly proportional to the brightness of the bass sound. On a rather soulful, Motown-esque bass track, for example, the net effect was subtle to the point of being unnoticeable. Only at the most extreme settings could I hear the C2 damping the signal. At that point, the results were not entirely pleasant. However, by backing off the input gain and adjusting the output volume to avoid distorting the signal, I was able to get a better result.

On the other hand, the C2 shone like a star when applied to a brighter track that included a Danelectro Longhorn bass. The compressor fattened up the bottom and middle frequencies without sacrificing the pick attack on each note—and it did it all in a perfectly transparent manner.

Next I auditioned the C2 as a direct box of sorts, recording various bass guitars through it and listening to the playback. The unit held up well, improving the sound of everything from an old fretless Fender Jazz Bass to a more modern graphite-neck model with active electronics. Again, though, there was a definite relationship between the high-frequency content of the original signal and the C2's effectiveness in improving the signal: the unit provided more immediate gratification on bright bass sounds, but it required extra tweaking to achieve good results from darker and more muted Motown-type timbres. Because of this tendency, I recommend that bass players strap on a new set of strings before recording through the C2.

Vocals and compression go together like Sonny and Cher.

I SING OF ICING

Vocals and compression go together like Sonny and Cher: it's a complex and difficult relationship with a long history, but the two are inextricably linked. Compression can smooth out an uneven vocal performance and tighten up background parts—or, in the wrong hands, it can rob a great vocal take of its dynamics. Of course, vocal compression is a matter of taste as well as of musical style, whether it's applied to triple-forte rock 'n' roll or a smoky torch song performed by a piano trio. So it was with some trepidation that I tried the C2 on this critical application.

Again, I found it easier to use the C2 to spruce up tracks after they were recorded, rather than using the unit to track with. I began by cutting my own glorious (read: erratic) voice onto an ADAT track with the C2 inserted into the path. I was curious to hear how well the unit functioned as a limiter, because one of the challenges in cutting vocals with a digital recorder is getting reasonable levels without clipping *and* without squashing the signal to an unusable degree.

In this case, the C2 required a lot of tweaking. I found myself wishing for a more predictable compressor—one with hard/soft-knee options and maybe even up-against-the-wall limiting. But to its credit, the C2 got the job done after I fussed with it for a while (and once I got accustomed to the way it colored the sound during the performance). Tracking with the unit requires a little patience; the C2's crankiness in this respect is a potential problem if you're trying to capture a performance quickly. But if time is not an issue, persistence combined with patience can yield gold.

For the second part of my vocal test, I ran some existing tracks (both lead and background vocals) through the C2. The tracks had already undergone some mild compression when they were cut, and the quality of the performances ranged from dynamic to uninspired. Here the C2 did amazing things. I was able to coax more life out of the performances, particularly on lead vocals that had plenty of attitude. Even on a more mannered vocal (or a voice-over, for that matter), the C2 was the icing that saved the cake.

KEYS TO SUCCESS

On keyboards in general, and acoustic piano in particular, I encountered some troubles with the C2. Most of the difficulty arose when I used it up front to compress or limit an acoustic piano

C2 Specifications

Audio Inputs	(2) ¼" balanced TRS (line-level)
Audio Outputs	(2) ¼" balanced TRS (line-level)
Attack Time	1–11 ms
Release Time	250 ms–3 seconds
Meter	LED; input level and gain reduction
Maximum Input Level	+28 dBu
Maximum Output Level	+28 dBu
Frequency Response	15 Hz–20 kHz (±1 dB)
Crosstalk	-50 dB
Distortion	0.01%
Noise	-90 dB
Signal-to-Noise Ratio	118 dB
Power Supply	12 VAC adapter
Dimensions	half rackspace x 5" (D)
Weight	2 lbs.

while recording. I employed a pair of Shure SM81 microphones and recorded the output to DAT with the C2 patched into the insert points of a small mixer. My goal was to tamp down the peaks mildly and warm up the overall sound as it went to tape, but I had a difficult time finding a setting that I could dial up and leave alone. I had to keep my eye on the C2's meters as I played and needed to stop constantly and review what I'd done. Even then, what went to tape sounded either over- or undercompressed.

The C2 was easiest to set up and sounded best when I used it to add detail to a continuous, Fats Domino–style rock part; it was less effective and harder to set for a two-part Bach invention. Still, by playing with the unit's Compression and Input knobs, I was able to dial up settings that were transparent yet yielded increased detail in both cases.

When I cranked the Compression knob to 11 and turned the input gain up fairly high, the C2 began to reveal its edgy character. At such extreme settings, it pumps and wheezes in a decidedly unfriendly manner, and the results are rarely usable. Nevertheless, I like the fact that the C2 can bare its teeth. You never know when such edginess might be just what a track needs.

On playback of a previously recorded piano part (on which I had used a high-end compressor sparingly while laying down the tracks), I found the going much easier. The C2 worked its magic, and I had a lot more fun tweaking the knobs. Finding settings that worked with the track was quick and easy, and I enjoyed auditioning them from the subtle to the extreme.

With synth patches, the results depended entirely on the patch itself. I treated a pizzicato string sample with some success: the C2 flattened and flattered the signal at the same time. The unit also worked well with the more percussive, Wurlitzer-type electric piano patches, brightening up the sound without making it seem forced or harsh. On patches with more sustain, however, such as a lush string sample or a Hammond organ with percussion, the C2 proved superfluous. I surmised that the rule to follow in this case is to compress only what is not already smooth in the first place.

JOEMEEK

C2 photo-optical stereo compressor

$399.99

FEATURES ■■■

EASE OF USE ■■■■■

AUDIO QUALITY ■■■■

VALUE ■■■■■

1 2 3 4 5

PROS: Easy to use. Portable. Great for guitars and vocals. Especially effective on individual tracks or stereo submixes.

CONS: Uses wall-wart power adapter. Can be temperamental when applied to final stereo mixes.

CIRCLE #442 ON READER SERVICE CARD

BANG THE DRUM

I love what the C2 does for drums and percussion. It behaved admirably for such a small box, producing sounds that rivaled those of far more expensive compressors, especially in the context of rock music.

When I applied the C2 to a stereo subgroup of drums and percussion pounding out a straight-ahead, four-on-the-floor rock beat, it performed some neat tricks. I especially liked how it handled the wash of the ride cymbals from a pair of left and right overhead drum tracks. The cymbal attack was compressed but not strangled, while the ensuing wash bloomed. This ability is not new, of course—compression has been applied to elicit this effect on myriad recordings—but the C2 did the job very well, especially considering its reasonable price.

The C2 responded nicely with the rest of the drum kit, too, and I was able to dial in a range of sounds that worked, from the snare on down to the kick drum. I did notice that the sound of the kit seemed to thin out if I overcompressed the tracks, but by backing off the amount of compression and adjusting the release time I was able to recover the heart of the beat.

I also recorded a few dumbek, tambourine, and bongo tracks. Again, good results took longer to attain when I cut tracks with the C2, but using it on playback was a total breeze. Most pleasing was the way it allowed me to bring up midrange detail without stealing the attack from each hit.

WHOLE ENCHILADA

Applying compression to a completed stereo mix can be a slippery slope even when you're working with the best of mixes and compressors. I found that the C2's effectiveness in this department depended on the type of music being compressed. When dealing with a "wall of sound" of guitars, bass, and drums pounding out a rock anthem, the C2 did remarkably well—I've spent more time tweaking much pricier compressors just to achieve the same result—but on sparser, jazzier arrangements with a bigger dynamic range, the unit was less effective (a more "tweakable" compressor would probably be called for in such performances). In either case, though, the C2 yielded improved stereo mixes.

THE VERDICT

The Joemeek C2 stereo compressor offers the smoothness, transparency, and seductiveness of photo-optical compression in a sturdy and portable package at a good price. In a world of computer plug-ins, complex multi-effects units, and user manuals the size of phone directories, it's nice to be able to sit down in near-total ignorance of a product and twiddle a few knobs to dramatic effect.

If you have some patience, the C2 can be coaxed to produce usable, often stellar, results in a wide range of musical applications. Overall, I found it more adept at processing individual tracks and stereo submix groups than final stereo mixes, and it is generally easier to use during mixdown than during tracking.

For the musician who is just getting started in the personal-studio biz, the C2 provides an excellent introduction to the world of vintage-style audio compression. Even if you already own a stack of high-end compressors, this little green box will prove itself a valuable and versatile asset to your audio arsenal. Likewise, for those with less-than-upscale studio budgets, the unit is a worthy and very affordable choice. And if you happen to fall into the guitarists-who-record-themselves-at-home category, I recommend putting the C2 at the top of your gear wish list.

John Ferenzik *plays keyboards and guitar with Todd Rundgren. When not on tour, he drinks a lot of coffee and records loud stuff in his basement.*

JAMES McCARTNEY

SUPERCOLLIDER 2.0 (MAC)

A sound-programming language for the next millennium.

By Thomas Wells

No matter how feature laden hardware synths and desktop music programs become, there comes a time when you bump up against these products' design limitations. With a sound-programming language such as James McCartney's SuperCollider 2.0, however, you can add nearly any function you want to your arsenal of music-making tools. Need a 40-operator FM sound? No problem. How about a 128-oscillator additive patch? That's easy enough.

There must be a catch, you say, and indeed there is. Instead of the familiar graphic interface and menu options found in modern desktop music programs, sound-programming languages present you with lines and lines of code that are used by the computer to create the sound you design. If you misspell a single word, the routine won't run, and if you forget to include even one value that a particular function needs, you also end up with nothing. Once you begin to work with such a language, however, you'll have incredible sound synthesis and compositional resources at your fingertips.

SuperCollider is an object-oriented programming language designed for software synthesis and signal processing in real time. It is fully MIDI integrated and virtually unlimited in its extensibility. The result of five years of development, SuperCollider is a remarkable achievement: it's fast, amazingly efficient, versatile, and expressive. It brings the benefits of modern programming languages such as SmallTalk and C++ to software synthesis (see the sidebar "Software Synthesis 101"). SuperCollider 2.0 is a major rewrite, not just a features-added rerelease of version 1.0. At the time of this writing, the current revision (and the subject of this review) is version 2.0.

GOING SOFT

Software synthesis uses a computer to generate waveforms that can be stored on disk or played back in real time. In the early days of software synthesis, the only "interface" was a computer terminal or card punch through which the synthesist/composer entered the lines of code that specified sounds and durations—not exactly the most efficient system for capturing musical ideas and performance gestures. With SuperCollider, a good bit of typing is still required, but you can also build your own graphic interfaces to control different parameters in real time, or even use your mouse to "perform" the sounds you've designed.

Many software synthesis languages are based on analog synth concepts and architecture, and the analog synth analogy is a good model for understanding SuperCollider. In place of the synth *module* however, synthesis languages like SuperCollider use *unit generators,* which are algorithms or functions that generate or process sound. Imagine all the different sound building blocks in your hardware synthesizer (filters, LFOs, oscillators, delays, and the like),

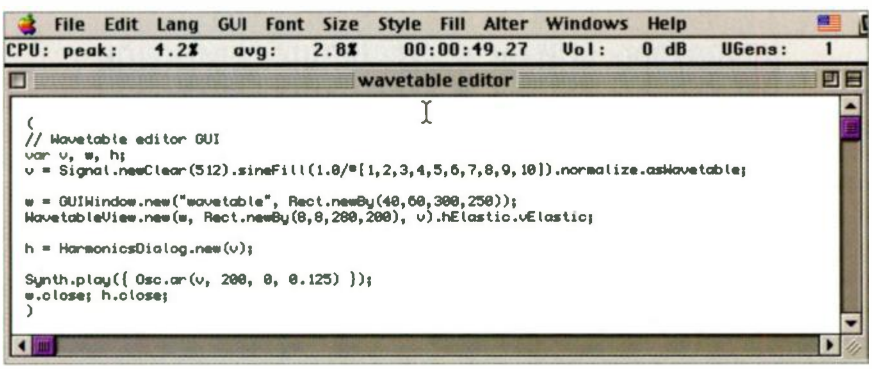

FIG. 1: As SuperCollider creates sounds, a status indicator at the top of the screen shows the amount of CPU resources being used.

and you'll have a good idea of the types of things you can use to build sounds with SuperCollider. Now add dozens of other sound-generating and sound-processing tools, mathematical functions for altering data, and the ability to patch together any of these functions in series or parallel. (See the sidebar "SuperCollider's Toolkit" for a summary of the available unit generators. A complete list can be found at the developer's Web site.)

Using SuperCollider, you build very complex synth "networks" by linking small components. In many cases, just a few lines of code can result in a very elaborate sound. You can easily generate an entire piece using this method, or you could create just a single sound of any duration. You could also transfer your SuperCollider sounds to a sampler or load them into a multitrack audio editor to assemble your final composition.

Of course, there are other sound-programming languages around today, most importantly Barry Vercoe's Csound. It, too, requires considerable typing and a good knowledge of modern synthesis techniques. But working with Csound compared with using SuperCollider is like programming in assembly language (in which every task must be broken down to its simplest components) compared with programming in a modern, high-level language. Csound does have the advantage of running on nearly any current computer platform, whereas SuperCollider is (for now) a Mac-only application. Csound is also a public-domain application and, as such, is available free of charge.

SHOW ME THE CODE

So what does a unit generator look like? How do you actually program a sound? We'll take a close look at two fairly simple examples and then cover some of the many functions SuperCollider provides. To get started, here's the code that would produce a 5-second sine wave at 800 Hz with a relative amplitude of 0.1:

```
(
Synth.play( { FSinOsc.ar( 800, 0.1 ) }, 5 );
)
```

The left parenthesis above and the right parenthesis below the code are strictly for convenience: in SuperCollider, clicking immediately to the

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The All-New JBL LSR Monitors are, quite literally, just that. Highlighted by a long list of performance-tailored components and customer-inspired features, they're like no other systems on the market today. The entire line, including the LSR32 3-way, 28P 2-way and 12P Subwoofer, is a technical triumph; resulting in new standards and performance levels for a rapidly emerging multi-channel recording industry.

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Revolutionary transducer designs, optimized network topologies and innovative materials are some of the reasons why the LSR line is being hailed as 'the world's most advanced monitor'. JBL's all-new ***Differential Drive***® woofer permanently dispels the notion that better linearity, higher power handling and greater dynamic accuracy are somehow an unobtainable, evil triangle. ***Dynamic braking*** produces truly accurate bass at higher SPL's with maximum reliability. Composite materials, including ***Carbon Fiber*** in the woofer as well as ***Titanium*** and ***Kevlar***® in the high and mid frequency components, insures performance that is always optimally maintained.

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While all companies boast about their specifications, JBL went one step further. To guarantee that every component of the LSR family worked together for optimal performance, LSR development employed JBL's unique 'system-engineered' design philosophy. Simply put: the entire line was researched and refined as one, with an overall performance goal in sight. What this means to you is a monitor and subwoofer that work together as a system; delivering stunningly uniform and accurate performance in both stereo and multi-channel applications.

TEC 1999 Nominee

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12" 3-way mid-field monitor with rotatable Mid/High Elements.

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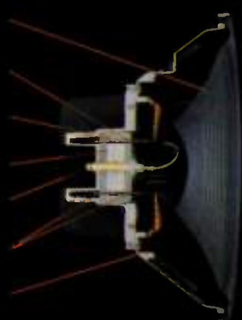

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To find out more about the revolutionary LSR Monitors, visit JBL Professional at www.jblpro.com granulation, and ring modulation, along with all the common synthesis methods in various forms. Each of these modules can be hacked easily and combined to build your own personalized SuperCollider processing and effects toolbox. (In this area, some type of graphic interface for building patches, such as those found in Opcode *Max* or Cycling '74 *MSP*, would be a big help to some users.)

You could press a single note on your keyboard controller that tells SuperCollider to start a multichannel algorithmic patch containing 10 or 15 layers of sound files (or several such patches simultaneously) with panning and other time-varying changes in the sound. Talk about "multisample groove machines"! This sort of material is right up SuperCollider's alley.

SuperCollider's facility to create copies of patches to incredible depths of duplication makes it a natural for basic sample-to-key mapping. Its processing speed (depending on your CPU, of course) lets you layer and transform sounds. You can also do multichannel work that is unheard of in hardware synths and would bog down many graphics-driven software-synthesis programs.

Of course, sample playback via MIDI key commands is nothing to write home about. But what if you could add morphing and other types of real-time spectral manipulation, and as many effects for each sample as you like, all with MIDI, mouse, or Wacom graphics tablet control? Those are all possibilities offered by SuperCollider.

To bring you back to Earth, SuperCollider does not have a turnkey sampling "module" (though you may find one in the many user-contributed examples). And to reach the level of mastery where you can build sounds like those described above definitely requires a large time investment. For me, however, SuperCollider radiates possibilities that are downright inspiring. Moreover, the SuperCollider package includes several intriguing examples that could be taken on the road or onstage with great effect.

SOUND CHECK

SuperCollider provides an impressive stock of audio and MIDI I/O options. I tested the program with a Korg 1212 I/O board as well as with Sound Manager and a Digidesign Audiomedia III card. The Sonorus StudI/O card is also supported.

The program worked flawlessly with the Korg board in all kinds of multichannel tests, and it worked similarly with Sound Manager. The Audiomedia III also worked successfully; in fact, all Digidesign cards are supported and have been verified by Digidesign to work with SuperCollider.

SuperCollider's MIDI-processing capabilities are extensive. The only problem I noted was a delay in real-time instrument triggering using Sound Manager I/O. This delay can be shortened, but only at the expense of general performance. With some MIDI-controlled patches, latency was still noticeable even with the Korg board.

SUPERCOLLIDER'S TOOLKIT

19 types of oscillators
36 unary operators (square root, transcendental functions, etc.)
31 binary operators (+, >, =, etc.)
14 types of noise generators
27 types of filters
19 types of controls
11 types of amplitude operators (limiters, compressors, etc.)
20 types of delays
FFT and IFFT
5 types of audio I/O
10 types of event spawning
5 miscellaneous unit generators

SOFTWARE SYNTHESIS 101

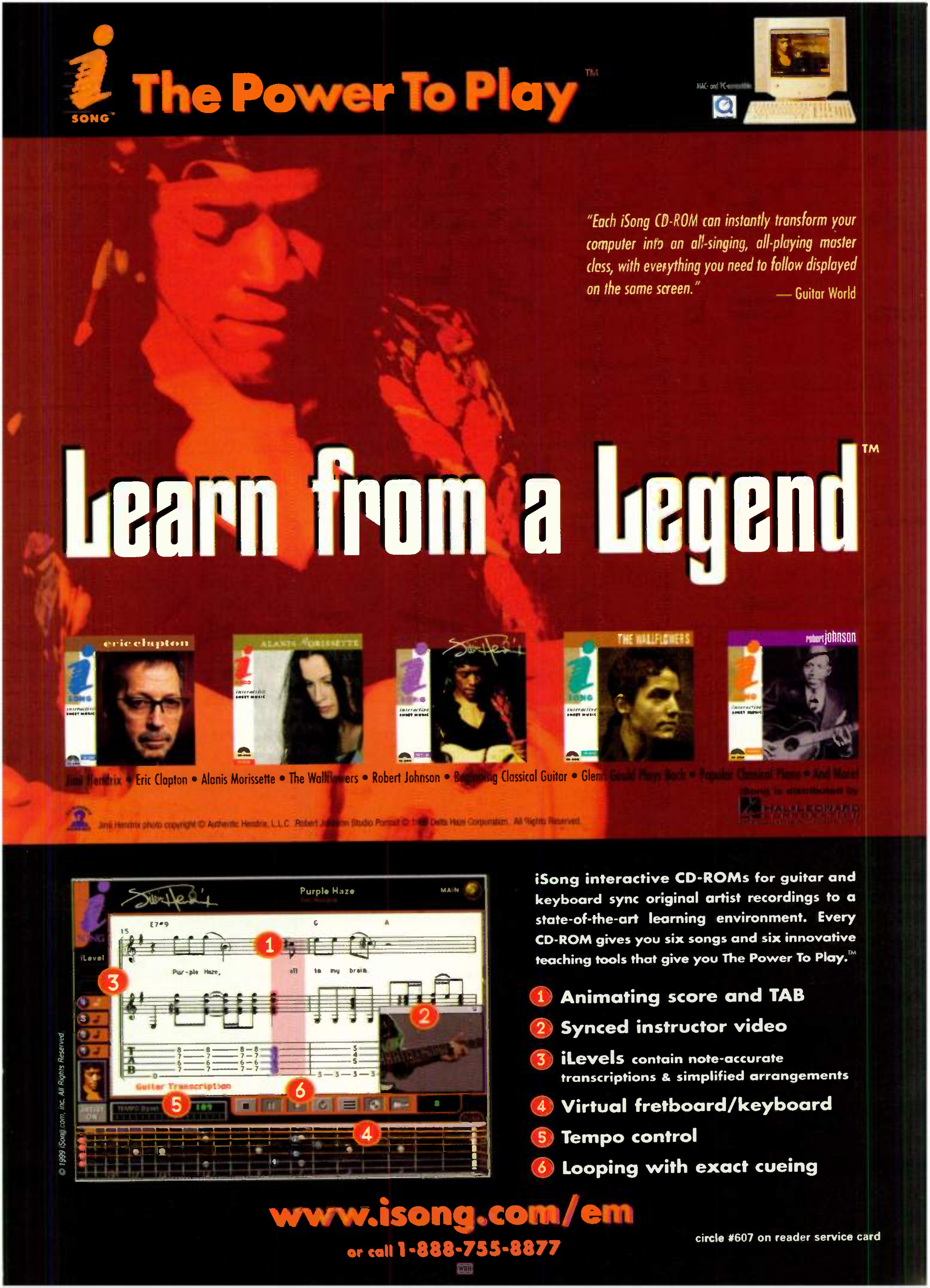
Computer sound generation has taken many forms, the earliest of which, software synthesis, was developed at Bell Laboratories in the late 1950s by the father of computer music, Max Matthews. Matthews documented his work in his book *The Technology of Computer Music* (MIT Press, 1969).

Matthews described a process by which a waveform could be calculated in very fine, evenly spaced time "slices" as a series of numerical values (samples), saved on some storage medium, and turned into sound by routing the stream of numbers through a digital-to-analog converter. Real-time software synthesis came into general use in the late 1980s with the advent of fast-enough computers.

Matthews recognized the need for a core library of programs and subroutines that performed the actions of oscillators, writing to disk, and other functions; he called these subroutines *unit generators*. Unit generators may be patched together to form what he called *orchestras*, or synthesis algorithms. The composer using particular unit generators had to memorize the arguments, usually a list of values separated by commas, that the generator required for setting pitch, loudness, phase, and other parameters. The output of a unit generator could be "patched" to other generators using variables that hold the value of the generator at each sample instant, giving the process a signal flow like that in an analog studio.

In the early days of software synthesis, limited computing power made possible only simple signal-generating and processing operations. As a result, many computer-music works from the 1960s and 1970s had a dry, unanimated, Hammond-organ-without-the-vibrato quality when compared with works produced in the analog studios of the time. Also, in the early days, sampling was rarely used because conversion equipment was unsophisticated and data storage capacity was limited. In addition, many of the synthesis algorithms we take for granted today had not been discovered.

Composers and researchers such as John Chowning, F. Richard Moore, Barry Vercoe, and many others throughout the '60s, '70s, and '80s created the software necessary for computer music to develop as an expressive medium. Nevertheless, in order to use such software, one had to write code.


iSONG
The Power To Play™
MAC- and PC-compatible
"Each iSong CD-ROM can instantly transform your computer into an all-singing, all-playing master class, with everything you need to follow displayed on the same screen." — Guitar World
Learn from a Legend™
eric clapton
ALANIS MORISSETTE
THE WALLFLOWERS
robert johnson
Eric Clapton • Alanis Morissette • Robert Johnson • Classical Guitar
Purple Haze
MAIN
iLevel
Guitar Transcription
iSong interactive CD-ROMs for guitar and keyboard sync original artist recordings to a state-of-the-art learning environment. Every CD-ROM gives you six songs and six innovative teaching tools that give you The Power To Play.™
1 Animating score and TAB
2 Synced instructor video
3 iLevels contain note-accurate transcriptions & simplified arrangements
4 Virtual fretboard/keyboard
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6 Looping with exact cueing
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Working with a sequencer running on the same computer is possible, but you could encounter problems if SuperCollider is stressing the CPU. Because the Mac OS does not allow the sequencer's timing interrupts to occur while the audio interrupt is happening, the sequencer could become choppy if SuperCollider's CPU usage is high. (Other real-time audio programs have the same problem on the Mac, so there's no reason to fault SuperCollider for this.)

SuperCollider has a few other limitations, as well. As powerful and robust a programming language as it is, SuperCollider doesn't yet offer some of the functions you find in other modern synthesis languages. For example, it doesn't have much in the way of physical modeling, and it doesn't yet have the range of spectral analysis and resynthesis tools you can find in a hardware-assisted package like Symbolic Sound's Kyma system. But these are minor concerns in a program so powerful, and you can be sure that many of these functions will show up in time.

In many cases, just a few lines of code can result in a very elaborate sound.

LEARNING BY EXAMPLE

When I first heard them more than a year ago, the highly interesting sound examples included with SuperCollider version 1.0 made a strong impression on me. Version 2.0 is even better in this respect. In a relatively short time, you should be able to find the significant parameters in many of these examples and create totally new sounds by changing them just a bit. You'll also enjoy "performing" many of the examples that can be controlled in real time by movements of your mouse (see **Fig. 4**). You'll also find a number of sounds in RealAudio format selected from the SuperCollider tutorial, as well as examples at www.sss.arts.ohio-state.edu, a Web site that is maintained by this author.

SuperCollider 1.0 received a tremendous boost with the appearance of a 135-page book written by veteran sound-synthesis author, programmer, and educator Steven Pope (available at www.create.ucsb.edu/htmls/sc.book.html). Unfortunately, Pope's examples won't run under the new version without some informed tweaking. A new, user-friendly, 2.0-compatible book by Alberto de Campo and Pope is in preparation, however; I have seen an advance copy, and I'm very impressed with its effective pedagogy and thoroughness of scope. (You can e-mail inquiries to de Campo at alberto@create.ucsb.edu.) Excerpts from the text are included free with the SuperCollider release.

You'll also find excellent help within the program. If you forget the arguments to a unit generator, SuperCollider offers ready assistance. For example, if you select SinOsc and type Command-H, you'll get very thorough and sympathetically written help in the form of text and working examples. If you forget which parenthesis or bracket balances which, you can go to the very end of the example, delete the right parenthesis or bracket, and retype it. SuperCollider highlights the complementary mark in the lines of code above. The program abounds with such useful debugging shortcuts.

ONE IN EVERY POT?

Will SuperCollider affect the price of potatoes in the general sound-synthesis

JAMES McCARTNEY
SuperCollider 2.0
sound-programming language
$250

FEATURES	4.5
EASE OF USE	3
QUALITY OF SOUNDS	5
VALUE	4.5
	1 2 3 4 5

PROS: Real-time performance capabilities. High-level language for sound synthesis. Rich library of 199 unit generators. MIDI integration.

CONS: Requires programming and sound-synthesis experience. Limited physical-modeling capability.

CIRCLE #443 ON READER SERVICE CARD

SuperCollider
Minimum System Requirements
150 MHz PowerPC CPU; 32 MB RAM,
Mac OS 8

community? Certainly, many of the program's beautifully worked-out details, its profound capabilities, and the fluidity of the language will be more apparent to those with the background to appreciate them, or who need them for their work. As McCartney puts it, "SuperCollider is probably not for beginners at programming or audio synthesis. It is easier to use than Csound but more technical than [Digidesign] *TurboSynth*."

On one level, users of Csound, cmusic, or cmix will feel comfortable with the general layout of SuperCollider and will appreciate its speed, brevity, and versatility. But SuperCollider goes far beyond a simple "modernization" of the Music *N* family. The integration of real-time synthesis with programming in a high-level language is an important and remarkable accomplishment. Certainly, if SuperCollider catches on, we will see archives of GUI-based SuperCollider applications comparable to those currently available for Csound.

At present, it's difficult to prognosticate about SuperCollider's future. I can't imagine a university or conservatory program in sound synthesis that would not include instruction in SuperCollider. For desktop composers who want to add a huge range of sound generating and processing options to their arsenal, it's one of the best bargains around. As of this writing, the sound-synthesis world is abuzz with things SuperCollider, and I believe it is rapidly becoming one of the premier software-synthesis programming environments. Will it eventually be ported to other platforms? McCartney won't make a commitment, but he says it's under consideration. For now, if you own a Macintosh, you can try out a sound-programming language that I feel certain will be around for a long time to come.

Thomas Wells *is a composer, author, and professor at Ohio State University. He has been involved with electronic music and audio production for more than 25 years.*

AKG

C 4000B

A versatile, fat-sounding large-diaphragm condenser mic.

By Karen Stackpole

On the heels of the SolidTube—AKG's most affordable tube mic to date—comes the company's next transducer event, the C 4000B, a large-diaphragm, side-address, electret-style condenser microphone. Similar to the SolidTube in design (the two mics employ a similar capsule), but with three polar patterns to the SolidTube's one, the solid-state C 4000B even looks like the SolidTube, except that the lower half of the mic is shorter. Of course, being solid state, the C 4000B doesn't have the SolidTube's little red "Easy Bake oven" window through which to see the glowing tube.

AKG bills the C 4000B as the "only electret, dual large-diaphragm transducer in the world." (What makes it unique is that its diaphragm is a full inch in diameter; evidently all other electret microphones on the market have smaller diaphragms.) Unlike "true" condenser mics, which rely on phantom power for the element's polarization charge, electret mics use a permanently charged element that does not require an external polarization voltage. Because the design eliminates the need for DC converters and other circuitry, electrets are less expensive to manufacture, and the savings can be passed on to the end user. However, electret-style condenser mics still need an external power source to operate the internal impedance converter.

I put the C 4000B through the paces in a number of studio applications and on a few location recording jobs, as well. Here's what I found out.

BUILD AND BOOTY

The C 4000B is built like a tank. Its all-metal housing gives the mic a satisfying heft and, according to AKG, extra abilities to deflect radio-frequency interference.

The mic's integrated wind/pop screen is also heavy-duty. I was impressed by the sturdiness of the wire mesh and foam that act as protective layers. They are so dense that, even when holding the mic up to a light, you can't see the outline of the capsule. The screen is very effective on its own, but just in case you need to record in a hurricane, an external foam windscreen is included.

You access the C 4000B's three polar patterns—cardioid, hypercardioid, and omnidirectional—through a recessed switch on the front of the mic. Two recessed switches on the rear activate a 10 dB attenuation pad and a 100 Hz low-cut filter. All electrical contact points, including the XLR connectors, are gold plated.

In addition to its foam windscreen, the C 4000B comes with a sturdy, spider-type shock-mount (model H100). It looks impressive, but when I gave the mount a shake, the rattling plastic pieces didn't exactly inspire confidence. The plastic threading in the mic-stand adapter also left me feeling dubious. However, once I inserted the C 4000B into the mount and tightened the dial that locks the mic into place, the rattling ceased and the mic was secure. Still, to avoid mishap, make sure you learn to use the tightening mechanism correctly.

Another little detail: the shock-mount has two different-size "strain-relief slots" that are designed to secure the mic cable and keep it from tugging on the mic. Unfortunately, the larger slot was too small to accommodate my Monster Cable.

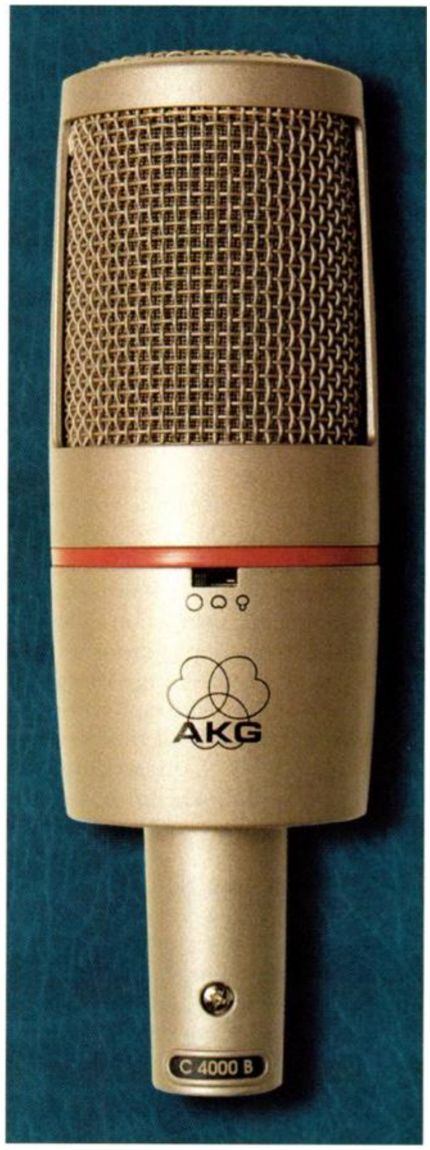
The C 4000B and accessories come securely nestled in a cardboard box lined with custom-cut foam rubber. Norbert Sobol, product marketing manager at AKG, told me that this is now the standard packaging for AKG's less expensive mics and was implemented to cut costs and reduce waste from discarded boxes. Sorry, folks—no snazzy aluminum flight case with this mic.

CHARACTER REFERENCE

To get a sense of how the C 4000B stacks up against its siblings, I first put it head-to-head with an AKG C 3000 ($438) and C 414 B/ULS ($1,285) on some standard loudspeaker tests. I also compared it with a more formidable contender in its price range; the Neumann TLM 103 ($995), to help draw a bead on the C 4000B's tonal character.

For the loudspeaker tests, I positioned the four mics closely together about 15 inches back from my stereo speakers and recorded three tracks—a folk song, a dub mix, and a big-band swing selection—to DAT with all the mics in the cardioid position. This test clearly revealed the response characteristics of the different mics.

The first thing I noticed was a definite high-end boost in the C 4000B. Compared with the C 3000, it seemed to emphasize frequencies between 6 and 8 kHz—which made the acoustic guitar and hats on the folk track seem overly bright. In some ways, the C 4000B's



A solid-state mic based on the design of the SolidTube, the AKG C 4000B is billed as the world's first electret condenser mic with a full 1-inch diaphragm.

Let the "critics" tell you how easy the Spirit Digital 328 mixer is to use...

Spirit's Digital 328 represents a new way of thinking in digital console design—it bridges the gap between analog ease-of-use and digital sound quality and features.

George Petersen of* Mix *says: "There are more than a dozen entries in the 'low cost' category of digital consoles, but in terms of pricing, performance and fast, logical interface, the Digital 328 clearly sets itself apart from the pack."

Take a few moments to read what he and other "critics" say about the Digital 328. Then, go to *www.spiritbysoundcraft.com* on the web for more information. If you're in the market for an *affordable* digital console, you need look no further.

On 328's user interface:

"The 328 is a real console interface that immediately feels as close to your comfortable old analog board as you could want... the consideration that has gone into every single button, knob and interconnect is striking." – Recording

"I liked the user interface a lot, and given that the most-requested features and digital interfaces are all included, the price is excellent." – Electronic Musician

"I like this board. It has a logical interface and enough knobs for fast operation (as such it could be ideal in a live performance or broadcast situation) while its audio performance is clean enough for any recording application." – Mix

On 328's E-Strip:

"The invention of the E-Strip is a stroke of genius, [giving] instant access to all controls at once on the selected channel." – Audio Media

"The 328 is fast and intuitive, thanks in large part to its 'E-Strip' interface. There are no subroutines or hidden pages; anyone familiar with an analog console can sit down at a 328 and be working in a matter of minutes." – Mix

"With Spirit's clever E-Strip design, this digital desk has the feel of an analogue." – The Mix (UK)

On 328's equalization:

"...To my ears, this is one of the most musical sounding digital EQs I've ever heard." – Recording

"[One] of the best features of the desk: carefully tailored to provide control ranges similar to those on a top-notch analogue console, it is (dare I say) very musical." – Audio Media

On 328's effects:

"A strong selling point for this unit is the pair of built-in stereo Lexicon effects... Having quality effects in the digital domain makes for clean sounds." – Electronic Musician

On 328's automation:

"The automation is straightforward to set up and works well." – Audio Media

"Between the user setups, snapshots and dynamic automation, the 328 remembers everything except the line-input trims and 100Hz rolloff switches. It's easy to get used to this way of working." – Electronic Musician

On 328's connectability:

"Clearly, the Digital 328 provides a multitude of configuration options suitable for project studios, post-production facilities, radio stations and even live applications." – Electronic Musician

"The 328 interfaces to practically anything digital." – Recording

On 328's unbeatable value:

"All in all, the British have indeed landed with a winner. The more you use this board, the more you will discover its depth and power. With one of these consoles, you could start a musical revolution of your own." – Electronic Musician

"This mixer packs a mighty punch for $5,000 [suggested list price]. It sounds excellent, does an excellent job of untangling all the various digital formats in use, and has an excellent interface. A bold step forward in digital console design." – Recording

"I like this desk! There's nothing better out there right now than the 328." – The Mix (UK)

On 328's mic preamps:

"The mic preamps have plenty of headroom... I was surprised at the clarity of the most subtle nuances of the percussion, including the last hint of sound from the bell trees and chimes." – Electronic Musician

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Rocklin, CA 95677
Toll Free: 888-459-0410

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"All in all, it is a delight to use—a real peach!" – Audio Media

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circle #612 on reader service card

highs were more akin to the C 414's, only crisper and not as smooth.

Also, compared with the C 3000, the C 4000B was fuller in the low end—again, similar to the C 414's. However, the C 3000 provided more midrange presence (around 3 kHz), which enhanced the clarity of the music. This was especially noticeable in the fuller orchestrations of the big-band selection and in the dub piece.

Evidently, the C 4000B's pumped-up lows and highs make its midrange response sound a bit distant—a quality all the more apparent in comparison with the TLM 103, which had smoother highs, clearer lows, and a better-represented midrange. (Of course, the TLM 103 offers neither an attenuation pad nor a low-cut filter, and it has only one polar pattern, which is cardioid.)

ON TRACK

Specs for the C 4000B are very impressive (see the table "C 4000B Specifications"). Self-noise and dynamic range, for example, are 8 and 137 dB, respectively, and the mic can handle sound-pressure levels up to 145 dB (155 dB with the 10 dB pad engaged).

I didn't stick the C 4000B into any motorcycle tailpipes to test its SPL handling, but I did test it on an electric-bass cabinet, a kick drum, and a variety of other percussion instruments—all without falter. I also tried the mic on hi-hat, acoustic upright bass, female and male vocals, and acoustic guitar. All tests were conducted using Focusrite Green Series preamps and Monster Cable Prolink 500 cables.

In general, the C 4000B proved to be a good performer, and it certainly impressed a few musicians on my location-recording jobs. It's a very quiet mic, and overall it sounded rich and crisp on the various individual sources I recorded. I especially liked it on shaker (miked from a distance of two feet), where it compared favorably to the TLM 103.

I also thought the C 4000B was very complementary to the vocal qualities of alto singer Heather Heliger, whose demo became a testing ground for the new mic. The C 4000B's full low end warmed up some aspects of her voice that might otherwise have sounded shrill. Although the mic's high-end boost sometimes resulted in too much sibilance, while its lows sounded just a tad muffled or covered, the overall sound was quite smooth.

On a mellow male vocalist already prone to sibilance and excessive mouth sounds ("spittiness"), the C 4000B was not ideal. Again, the sharp highs only enhanced the spittiness, and the lows came across sounding slightly covered. Thanks to its full lows and tendency to downplay mids, this mic would probably work best for a vocalist who has a biting, midrange-type voice.

DRUM DELIVERY

I got excellent results from the C 4000B on dumbek—a good instrument for testing a mic's response, thanks to its unique balance of highs, lows, and mids. The dumbek I used had a fundamental note of low B, which is around 246 Hz. The C 4000B emphasized this tone nicely, confirming my general perception of the mic's characteristic boost around 250 Hz. The C 4000B's boosting of high frequencies was favorable for the dumbek, too, resulting in a nice "papery" sound with plenty of attack and detail from the head and fingers. Transient response was also good, though not exceptional.

For manufacturer information, please see Contact Sheet, p. 179.

In another comparison test, I close-miked a double-headed bass drum (no hole in the front head) with the C 4000B and the C 414, both in omni mode and with 10 dB pads engaged. The performance of the two mics was similar, except that the C 4000B sounded thicker and captured a bit less attack. Still, this was one of my favorite applications for the mic.

Next, I moved both mics back five feet and raised them several feet for a distant-miking test on the same drum set. Here I found that the C 4000B colored the sound of the hi-hats noticeably, resulting in a less realistic sound than what the C 414 captured. Specifically, the C 4000B accentuated a high, clangy ring while simultaneously attenuating some of the lower overtones. It wasn't a bad sound (and it was less clangy than what the C 3000 picked up), but it did demonstrate the particular high-end boosting characteristic of the mic.

As for the kick-drum sound picked up by the distant mics, the lows from the C 4000B sounded a bit too thick and muffled for my tastes. The effect was sufficiently bright but there was a lack of clarity, caused, apparently, by the slightly muffled low end.

STRING THING

The C 4000B sounded rich and full on a bass cabinet during a Latin jazz session in which the bassist played an electric upright. This, too, ended up being one of my favorite applications for the mic. On acoustic double bass, however, it accentuated an annoying buzz from the strings. Although the C 4000B gave a fairly accurate representation of the instrument, the troublesome buzzing remained, even though I tried several mic positions.

AKG
C 4000B large-diaphragm electret condenser microphone
$848

AUDIO QUALITY ■■■■
VALUE ■■■■
1 2 3 4 5

PROS: Well featured. Quiet. Full low end. Big dynamic range. Handles high SPLs. Comes with shock-mount and external windscreen.

CONS: Somewhat muffled low-end response. Accentuates sibilance on certain vocalists.

CIRCLE #444 ON READER SERVICE CARD

The C 4000B proved well suited to a steel-string acoustic guitar. The mic's high-end rise made for a bright, sparkly sound that would help the track cut through a mix. However, the overall sound was again somewhat muddled by the C 4000B's thick, low-mid boosting.

ON THE CHARTS

During the loudspeaker tests, I also listened to the sonic differences in the C 4000B's polar patterns. The selector switch was easy to manipulate, although reaching it was a little tricky when the mic was in the shock-mount. I was pleased that the sound of flipping the switch was hard to hear in the speakers.

In cardioid mode, the mic's high- and low-end responses increased and the bass response became noticeably colored. This boosting correlates with AKG's published frequency-response charts, which show peaks at 4 and 10 kHz.

In the mic's hypercardioid mode, the midrange became more distant and "pinched" sounding, while the bass dropped off considerably. Also, off-axis response grew more colored (typical for a tight, directional pattern), and the sound seemed slightly compressed. Again, what I heard jibed with AKG's specs, which show a rolloff in the lows below 100 Hz and some dips in the mids between 1.5 and 3 kHz. Also, the highs were slightly less bumped up in hypercardioid mode (as compared to cardioid), though still plenty bright.

In omni mode, the mic's hyped lows and highs evened out nicely. The midrange around 3 kHz came up, too, resulting in a more accurate, open, and natural sound. This was my favorite polar pattern on the C 4000B.

C 4000B Specifications

Acoustic Operating Principle	pressure-gradient transducer
Element	fixed-charge backplate; permanently polarized condenser (electret)
Diaphragm	1", 6-micron-thick, gold-vapor-deposited Mylar
Polar Patterns	cardioid; hypercardioid; omnidirectional
Frequency Range	20 Hz–20 kHz
Dynamic Range	137 dB (A weighted)
Signal-to-Noise Ratio	86 dB
Sensitivity (@ 1 kHz)	25 mV/Pa
Self-Noise	8 dB (A weighted)
Maximum SPL (for 0.5% THD)	145 dB (155 dB with 10 dB pad)
Highpass Filter	100 Hz; 12 dB/octave
Attenuation Pad	10 dB
Dimensions	7.2" (L) x 2.3" (D)
Weight	1 lb.

ONE FOR ALL

The C 4000B is a versatile, fat-sounding, and affordable large-diaphragm condenser mic. In general, it exhibits crisp highs (especially around 6 to 8 kHz) and a very full low end that is similar to the AKG C 414's but with a noticeable boost around 250 Hz. The audible emphasis on the lows and highs can make the midrange sound distant, and the highs, while crisp, sometimes exhibit a slight lack of definition. But I'm splitting hairs here: this mic's sound ranged from good to very good in all my tests.

The C 4000B is also remarkably quiet, is capable of handling high SPLs, and comes with a lot of extras, including a 10 dB pad, a 100 Hz rumble filter, and three polar patterns. These qualities and features, along with the inclusion of a quality shock-mount, make the mic a very good value. Overall, the C 4000B is a great choice for the personal- or project-studio owner seeking a versatile, low-cost, and high-quality condenser mic that just about does it all.

Karen Stackpole *is a recording and mastering engineer as well as an active drummer/percussionist. She wanders the San Francisco Bay Area under the alias Stray Dog Recording Services.*

ADVERTISER INDEX

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	Very Helpful	Somewhat Helpful	Not Helpful	Didn't Read
A. "Dance of the Dead," p. 30	701	702	703	704
B. Cover Story: "The Path of Least Resistance," p. 44	705	706	707	708
C. Production Values: "Diva's Choice," p. 72	709	710	711	712
D. Square One: "Picture Perfect Sound," p. 96	713	714	715	716
E. Working Musician: "Expose Yourself," p. 106	717	718	719	720
F. Final Mix: "Information, Please" p. 210	721	722	723	724

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Cover Story: The Path of Least Resistance
pp. 44–70

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PreSonus Audio Electronics tel. (800) 750-0323 or (225) 344-7887; fax (225) 344-8881; e-mail presonus@presonus.com; Web www.presonus.com

Rane Corporation tel. (425) 355-6000; fax (425) 347-7757; Web www.rane.com

Reviews
pp. 114–188

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Alesis Corporation tel. (800) 525-3747 or (310) 255-3400; fax (310) 255-3401; e-mail alecorp@alesis1.usa.com; Web www.alesis.com

Beatboy tel. (800) 838-BEAT or (570) 685-1338; fax (570) 685-1573; e-mail beatboytec@aol.com; Web www.beatboy.com

CreamWare US, Inc. tel. (800) 899-1939 or (604) 435-0540; fax (604) 435-9937; e-mail info@creamware.com; Web www.creamware.com

E-mu–Ensoniq tel. (831) 438-1921; fax (831) 438-8612; e-mail info@emu.com; Web www.emu-ensoniq.com

Frontier Design Group tel. (800) 928-3236 or (603) 448-6283; fax (603) 448-6398; e-mail info@frontierdesign.com; Web www.frontierdesign.com

James McCartney e-mail info@audiosynth.com; Web www.audiosynth.com

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TC ELECTRONIC

M3000

Big-time effects with a friendly interface.

By Mike Collins

TC Electronic has a history of creating serious effects processors, ranging from a studio-quality stompbox chorus all the way to the high-powered (and high-priced) M5000. Its M2000 Wizard has become a staple in many personal studios, where it handles a variety of processing chores with aplomb.

In the new M3000, the Danish manufacturer has gone a step beyond the M2000; this high-quality reverb and multi-effects unit features 24-bit A/D and D/A converters and newly developed reverb algorithms that rival those of the best processors around. The unit has two separate processors, which can be routed in Serial, Parallel, Dual Input, Dual Mono, Linked, or Preset Glide modes.

PAUSE TO REFLECT

There's no lack of patch memory, with 250 single and 50 combined presets in ROM and the same number in user RAM. You can store the same number of patches on a standard PC Card.

The reverb presets run the gamut; there are halls, rooms, plates, and gated reverbs, as well as various specialized presets for post-production applications. The M3000 also offers delay, pitch, EQ, expander, chorus/flange, compressor, tremolo/pan, phase, and de-esser effects. Several preset combinations show off some of the more complex effects you can achieve using both processors.

According to TC Electronic, the new reverb algorithms are the result of the company's Virtual Space Simulation (VSS) research program. Because TC's research shows that no pitch modulation occurs in natural reverb, the goal of the VSS program was to design a reverb that wouldn't add modulation to its diffusion system. The result is a greatly enhanced stereo image. However, pitch modulation provides a lot of character for some classic reverb sounds, so the M3000 offers both "natural," modulation-free presets and "vintage" presets with pitch modulation.

The M3000 also benefits from the VSS team's improvements in reflection emulation. M3000 reverbs offer eight discrete reflections, combined across the L/R outputs. Each of these reflections has individual EQ, delay, level, and phase settings. The VSS research program is ongoing, and TC is currently developing the next set of algorithms, which will be targeted specifically at post-production applications. All of the VSS algorithms being developed for the M3000 will be ported over to the M5000.

STYLISH LOOK

Although the M3000 sports a slightly different look than other TC Electronic boxes, the basic layout is consistent with previous designs. The M3000 is a 1U rack-mount device with a PC Card slot on the left-hand side of the front panel and an on/off switch above that. There is a display area to the right of the card slot, with a pair of input LEDs, several overload/sample rate LEDs, and an LCD parameter display screen. To the right of the display are 24 buttons, laid out in six columns of four, which provide access to the parameters. An infinitely rotating Adjust (data-entry) wheel graces the far right of the front panel.

On the back panel are an additional power switch and a power socket, a pair of balanced XLR inputs and outputs, S/PDIF I/O on both RCA and optical connectors (the optical ports also can handle ADAT Lightpipe signals), word clock I/O on an RCA connector (an RCA-to-BNC adapter is supplied), and XLR jacks for AES/EBU I/O. MIDI In, Out, and Thru ports are provided, as is a ¼-inch jack for connecting a momentary footswitch. You can use the pedal for tap tempo or as a bypass switch for either (or both) of the processing engines. All the audio connectors are gold plated, and any or all of the digital outputs can be used simultaneously.

EASY NAVIGATION

The first column of buttons provides access to the M3000's global utility pages. Here is where you configure the unit's inputs and outputs, set routings (more on this later), adjust the I/O levels, and control various utility and MIDI functions. Effects parameters are set using three columns of Recall, Store, Bypass, and Edit buttons. Because the M3000 contains two separate processors, you can configure the two processing engines independently, or you can do them in combination so that a single set of controls is employed.

The fifth column of buttons provides four snapshot locations for quickly storing and recalling the combined presets, and the final column is reserved for control functions: an OK button to confirm operations, a Shift button to let you access secondary functions, and a pair of cursor keys for moving between parameters when editing. If you press the Shift key first, hitting OK cancels you out of the current operation. Likewise, a combination of the Shift and Up cursor keys takes you to the top of the currently displayed parameter list, while pressing Shift and the Down cursor takes you to the bottom of the list.

To recall a program, simply hit the Recall key and scroll through the presets using the Adjust wheel. Because of the number of presets, you could be doing this for a very long time. But don't worry: TC Electronic has provided an index screen that can be accessed by holding the Recall key for a few seconds. This display reminds you which preset types are stored where; for instance, you can see at a glance that the Gated Reverbs presets are numbered from 205 to 211.

The TC Electronic M3000 offers a wealth of features, an excellent user interface, plenty of effects algorithms, and reverbs that compare with those on units that cost twice as much.

Somebody's Been Thinking About You

In designing our new workstation we had to carefully consider many things, but above all, the most important was what <u>you</u> might actually want. So if you're about to settle for the usual mix of synth and sequencer with a few extra frills thrown in; wait ! Wouldn't you like to have a top of the line pro piano too? What about a drawbar organ? Or how about a universal sample player? Or a groove machine? The Equinox Pro is all of these things and much, much more at an unbelievable price that makes sweet dreams come true. Take a look...

SOUND LIBRARY - Nearly 1200 sounds "out of the box", ranging from the phattest analog synth patches to crisp acoustic and orchestral tones, and without having to buy any additional ROM modules. These sounds, created by some of the top programmers in the U.S. can be quickly and easily edited and over 2000 user sounds can be saved into the standard on-board memory.

THE REAL PIANO* - Generalmusic's PRO series physical modeling pianos hardly need any introduction at all. An established favorite of many of the world's most prominent recording and touring pianists, the Grand Piano, Rhodex, Wurli, CP and Clavinet sounds from the, "top of the line" PRO2, are included as standard on the Equinox PRO along with our patented "Damper Physical model" and "Natural string resonance" technologies.

PERFORMANCE - Combine up to 16 sounds together in any kind of multi-split or layer configuration. The sliders and buttons can instantly be used as mutes and volume controls for your 16 parts.

DRAWBARS - One innocent looking little button transforms the Equinox PRO into a versatile drawbar organ. While the sliders become a set of traditional organ drawbars, the front panel offers instant access to Key Click, Percussion and Rotary speaker speed, (all user editable). There's even a choice of Drawbar voicings, (Smooth, Hard, Jazz and Rock) and the added flexibility to edit each drawbar's pitch, and pan position individually.

REAL TIME CONTROLLERS - Whether you're playing single sounds, performances, samples, grooves or sequencer songs, grabbing a handful of sliders will instantly transport you to analog heaven, (and without the need for WD40). Sliders are pre-assigned to control envelope attack/ decay/ release, filter cut-off / resonance and LFO depth and speed but any slider can easily be re-programmed to control the parameter of your choice. In sequencer mode the sliders can also be set to function as a 16 track mixer with the press of a single button.

DRUMS KITS - 43 on board drum kits from a brand new sample library ranging from hip-hop, dance and techno to fusion, rock, jazz and 70s. Editing is quick and easy and you can save as many user drum kits as you like, (up to 2048 kits).

USER PANEL - Each of the 16 sliders and 16 buttons, (8 physical buttons and sliders with a SHIFT key), can be individually programmed to control either the internal functions of the synth engine and sequencer or they can be set up to send any MIDI controller, system common or system real-time message of your choice to either of the two MIDI out ports. Up to 16 complete panel configurations can be saved and independently recalled.

4 INDEPENDENTLY ASSIGNABLE OUTPUTS - Each performance part, sequencer track or drumkit instrument can be individually assigned to any of these outputs either as a stereo pair or as a single mono out.

SINGLE INSTRUMENT GROOVES - Any musician will find our collection of "single instrument grooves" indispensable. These superb phrases, offering everything from drum loops and bass lines to clavinet riffs, organ licks, guitar grooves and a whole lot more, have been created by some of the worlds top session musicians. And they're yours to use as you please in any sequence or song creation of your own. We've already taken care of the royalty payments !

CONTROLLER/6 MIDI PORTS - With no less than 6 independent MIDI ports, the Equinox PRO offers full control over 32 channels of MIDI data. The keyboard can be used as a 16 zone controller with each zone having its own programmable upper and lower split points and velocity limits, (which may overlap with other zones - of course !)

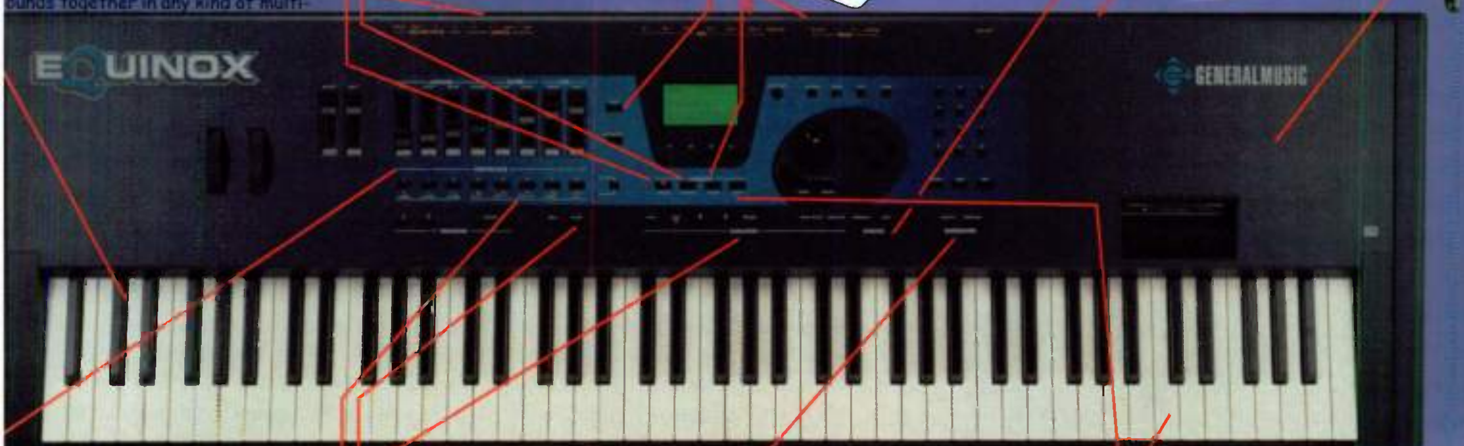
3 PROGRAMMABLE PEDAL JACKS

DJ GROOVES - Having 1024 on-board grooves puts the Equinox PRO head and shoulders above any of those stand alone "Groove Stations". All grooves can be edited, filtered, remixed and even "scratched DJ style" in real time. Plus you can create your own grooves from any midi file or sequencer song.

HARD DISK* - The Equinox PRO comes as standard with a 2Gigabyte internal hard disk. Anything can be stored on the hard drive - Sounds, Performances, Drum Kits, Imported Samples, Grooves, Midi Files etc. The hard disk can also be accessed "invisibly" so that new songs can be loaded while the sequencer is already busy playing another song.

SCSI INTERFACE*

EQUINOX PRO



POWERFUL SEQUENCER - The versatile 16 track sequencer offers a staggering 250,000 events of storage and allows you to store 16 songs in memory at once. Add to this the power of 1/192 resolution, Groove quantize, Event list editing and the ability to record ALL slider movements, (either real-time editing of filters, envelopes etc or multiple track mixdowns) plus the life saving UNDO function and you'll start to see why the Equinox PRO is really a composer's dream machine. The sequencer even has it's own 16 tracks, independent from the rest of the synthesizer so, while a 16 track song is playing, you can freely select any complex multi-split or layered performance to play along with it. Simply put, it's just like having two separate keyboards.

EFFECTS - Up to four independently assignable DSPs are available offering 85 crystal clear digital effects. Ranging from simple reverbs and choruses to complex composite algorithms like GUITAR FX, (Distortion, Gate, Delay and 4 Band EQ using only one DSP section !), 3D ENHANCER, AUDIO EXCITER, RING MODULATOR, 10 BAND EQ and 4 PART PITCH SHIFTER.

ARPEGGIATOR - With 16 factory presets and 16 user programmable patterns the Equinox PRO's Arpeggiator will send you soaring back to the 70's, (with a few little technological miracles thrown in for good measure). As well as the usual UP and DOWN directions there's also RANDOM and INPUT, (the order in which the notes were played). Velocity can be pre-set, disabled or controlled by your playing style and there's also a CRESCENDO feature to reduce or increase velocities as the pattern cycles around. There's even a HOLD button which lets your pattern do it's thing while giving you both hands free to tweak the arpeggiating sound with the real time controllers.

88 NOTE PIANO ACTION - Fully weighted 88 key hammer action gives you the authentic feel and response of a fine acoustic piano.

SAMPLE TRANSLATOR - The Equinox PRO comes standard with 8Mbytes of Sample RAM, (expandable to 40Mbytes), which can be used to load samples in Akai, Kurzweil, WAV and AIFF formats from CD-ROMS or floppy disks, (with other formats to be supported in future o.s releases). Loaded samples can be freely edited like any other sound and they even stay in memory after you switch off the power.

<u>Breakthrough Price: $ 3,695.00</u>

Also available: Equinox 61 $ 1,995.00
Equinox 76 $ 2,195.00

* Optional on Equinox 61 & 76 models

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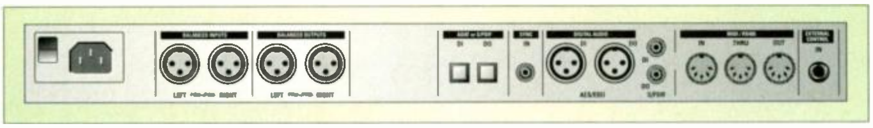


In addition to balanced analog I/O, the M3000 offers AES/EBU and S/PDIF coax and optical I/O. The latter can also handle ADAT Optical I/O.

The M3000 offers a really neat feature called the Recall Wizard. The general concept is that you can pick an application type (such as music or post-production), choose the kind of audio source material you are working with (drums or guitars, for example), and then decide what size acoustic space you want. Once you have specified these criteria, the M3000 presents you with a list of presets (typically between 5 and 20) that should fit your needs. This saves you the trouble of auditioning hundreds of presets to find the ones that fit the bill.

Hitting an Edit button brings up a scrollable page that contains the main parameters of that engine: decay time, wet/dry balance (Mix), and output level. In the case of the combined presets, there are only two editable parameters: the first lets you balance the output of the two processing engines, and the second allows you to set up the Dynamic Morphing (more on this in a moment).

An Expert editing mode is available for the new VSS algorithms, and it provides many more parameters. These algorithms have separate parameters for the reverb (such as decay time and mix level), the early reflections (type, balance, and so forth), the reverb tail (high, mid, and low decay times, for example), modulation of the reverb tail, and Space Modulation, which controls the way the sound moves within the room. If you simply want to set the reverb decay time and tweak one or two other settings, the M3000 couldn't be easier to use. If you are seriously into sound design and want to create your own megareverb presets, Expert mode awaits you.

TWICE AS NICE

The beauty of having two processing engines in one unit is obvious, and after working with the M3000 for a while you begin to realize just how flexible this device is. Think of the box as being two separate stereo units that can be linked together in various ways.

With both processors operating in stereo, you can feed the output of one effect to the input of the other; TC Electronic refers to this as Serial mode. In contrast, Parallel mode lets you input a signal from either the left channel (for mono operation) or both channels (for stereo operation) and mix the outputs of both processing engines into one stereo signal. Keep in mind that there are only two physical channels for input and output; if you want to use the effects completely

M3000 Specifications

Analog Inputs/Outputs	(2) XLR/(2) XLR
Digital Inputs/Outputs	(1 pr.) AES/EBU (XLR); (1 pr.) S/PDIF (RCA); (1 pr.) Toslink optical (ADAT or S/PDIF)
Other Connections	(1) ¼" footpedal, (1) RCA word-clock I/O, (1) PC Card slot
Preset Locations (ROM/RAM)	300/300
A/D/A Conversion	24 bit
Frequency Response	10 Hz–20 kHz (+0/-0.5 dB)
Dynamic Range	>100 dB (unweighted) >104 dB (A weighted)
Total Harmonic Distortion	-95 dB (0.0018%) @ 1 kHz
Sampling Rates	44.1 kHz, 48 kHz
Processing Types	reverb, delay, pitch shifter, EQ, expander, compressor, chorus/flange, tremolo/pan, phase, and de-esser
Dimensions	1U x 8.2" (D)
Weight	5 lbs., 3 oz.

For manufacturer information, please see Contact Sheet, p. 179.

independently, you will have to use the Dual Mono mode, which uses the left-channel I/O for Engine 1 and the right-channel I/O for Engine 2.

Dual Input mode operates similarly, with the left input feeding Engine 1 and the right input feeding Engine 2. However, in this case the outputs of the two engines are combined in stereo at the output; you can mix them as you wish. This mode is good for situations in which you need to use a couple of effects units but have only one pair of effects returns available. Of course, you might simply want to use the M3000 as a single stereo effects unit, so there's also Linked mode, in which the two engines act as one with their edit pages locked together. In this mode, both sets of parameters contain the same settings, and the audio passing through both processors is phase-locked together.

One of the really cool things about the M3000 is the Preset Glide Mode, which allows seamless transitions between effects. By setting a crossfade from one preset to another and adjusting the fade time, you can switch programs without hearing any glitches or silence from the M3000. Creative possibilities abound: for instance, you could allow a delay to keep repeating while you're fading into a chorus.

Dynamic Morphing is similar to Preset Glide but works in Parallel mode and fades between parameters within a single preset. The morphing between parameters occurs based on a threshold you set for the input signal. For example, you could use this to increase the room size of a reverb for a song's chorus: if the vocalist sings softly in the verse and louder in the chorus, you can set the input's threshold so that when the chorus kicks in, it triggers the morph. You can also specify how long it takes to morph from one effect to another (slow, medium, or fast).

The beauty of having two processing engines is obvious.

CONCLUSIONS

The M3000 would be a fine addition to most processing racks. Its user interface is extremely easy to learn and to use, and the unit is packed with features that will make your recording life easier. You can alter the level of the digital input to compensate for a DAT that was recorded too low, toggle the bit rate of the digital output between professional and consumer formats, or dither the digital output from 24 bits down to lower resolutions (as far down as 8 bits). Another neat touch is the MIDI Monitor window, which lets you display incoming Program Changes, Control Changes, SysEx data, and Note On/Off messages. The only thing I didn't like about the M3000 was the LCD screen, which I felt was too small and too difficult to read from an angle. Also, the unit's lack of ¼-inch analog I/O would be inconvenient for many personal studios.

The M3000 has a wider range of effects than many of its competitors. Sure, the older reverb algorithms taken from the M2000 and M5000 are good to have, and the rest of the multi-effects are useful enough, but the new VSS algorithms are what make the M3000 stand apart from the competition. They provide a new level of realism and offer more control than any other product out there. Quite simply, this is the most natural-sounding digital reverb I have ever heard.

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

TC ELECTRONIC
M3000 multi-effects processor
$2,495

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

PROS: Amazing reverbs. Easy to use. Two discrete effects engines. Wide range of effects.
CONS: Small LCD can be awkward to work with in Expert mode. No ¼-inch analog I/O.
CIRCLE #445 ON READER SERVICE CARD

QUICK PICKS

WORLD WIDE WOODSHED

SlowGold II 5.1.2 (Win)

By Mike Lawson

World Wide Woodshed has released an update of *SlowGold II* ($49.95), a program that allows you to capture and slow down music without changing the pitch so you can learn or transcribe parts. The program works with WAV files, and it can record from your audio CD in either digital or analog mode, depending on what your CD-ROM drive supports. If you have audio files in MP3 or other formats, you will have to convert them to WAV format before bringing them into *SlowGold II*.

The program gives you a lot of options. You can set loop points in the selection using a waveform view in a pop-up window. Loops will play back at the default 50 percent speed or the playback speed you last used, or you can reset playback to any speed between 5 and 120 percent. You can choose to sacrifice sound quality if you want faster time-stretch processing, or you can opt for crystal clear quality if you have either a fast machine or a lot of patience.

SlowGold II allows you to build a loop database that includes chords, lyrics, and comments for future practice sessions. Another useful feature is the unique Rhythm Grid, which allows you to see the rhythm of any passage by placing a beat "ruler" over the waveform of your loop.

Copping Chops in the Woodshed

Having released a CD last year that featured several top-notch guitarists—Jorma Kaukonen, Bob Welch, Joe Louis Walker, and others—I was quite eager to try this software so I could attempt to play the solos they contributed. The installation was effortless, and I immediately set about finding edit points for loops from the tracks I wanted to learn.

SlowGold II is pretty self-explanatory, so I had no trouble getting started. I selected track 1 of the disc, set the edit points for Kaukonen's solo, and allowed *SlowGold II* to record the loop to my hard disk. Within minutes my loop played back to me at half speed, sounding as though the band had eaten a handful of Quaaludes. I grabbed my guitar and began to hunt and peck for the starting note. After a few passes I was able to play the solo at the full speed of the original recording. Success!

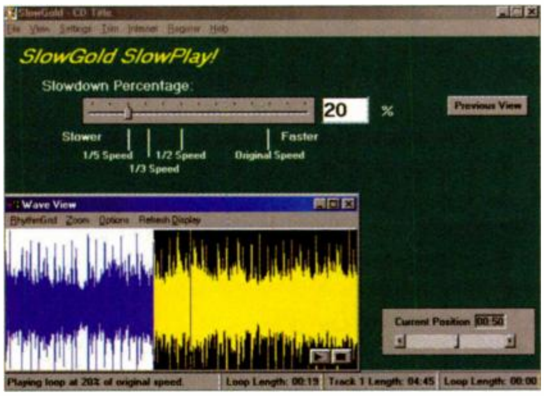
The first solo I looped into *SlowGold II* was a short one—about 15 seconds long, so I decided to try it out with a longer passage. I set new loop points and recorded a 45-second solo section to my hard disk. This time, there was a noticeable wait for the loop to change speeds—about two minutes. It took this long each time I wanted to slow the section down again. (With version 6.0, which may be available by the time you read this, World Wide Woodshed plans to offer real-time playback of slowed-down files, as well as independent pitch shifting.)

Up to Speed

SlowGold II can run under Windows 95, 98, or NT, and it requires at least 12 MB of RAM and 16 MB of available hard-disk space. You'll need at least a 80486/66 MHz PC, although a 166 MHz Pentium or better is recommended. I tested it on a Pentium II/350 running Windows 98 with 256 MB of RAM and found that it performed very well, even though it took a couple minutes to process longer loops. That's not too bad with a fast machine, but I have to wonder whether it would perform very well on a slower system unless the user were willing to opt for the poorer audio quality.

In the guitar world, it seems like everyone is marketing "slow down" products of one kind or another, from the old Ibanez multispeed cassette deck to multi-effects pedal boards (like the DigiTech RP7). Many manufacturers have recognized the educational value of being able to time-stretch samples of musical parts. If I am seriously attempting to learn a part, though, I would rather sit down in front of my computer with *SlowGold II* than plug my CD player into a foot pedal and try the same thing. Having visual control over the edit points of the loops is a tremendous benefit.

I intend to keep working with the software to learn more of the solos from my guest guitarists. I only wish it could help me get used to wearing these dang finger picks so I could play Kaukonen's parts correctly.



World Wide Woodshed's *SlowGold II* provides easy edit-point control for slowing down WAV or audio files.

Overall EM Rating (1 through 5): 4

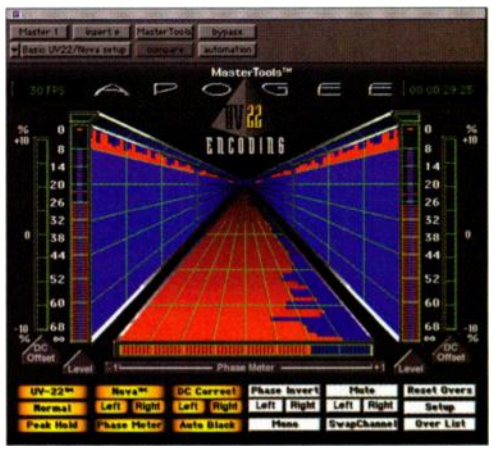
CIRCLE #446 ON READER SERVICE CARD

STEINBERG

MasterTools 1.5 (Mac)

By Mikail Graham

When Apogee first introduced *MasterTools* to the TDM world in 1995, the company aimed to provide a host of pro-level mastering features—phase inversion, mono-compatibility monitoring, DC offset correction, and variable scale display meters, to name a few—in an affordable and easy-to-use software plug-in package. Unfortunately, *MasterTools* 1.0 required that a pair of administration programs run in the background, which often caused problems.



Originally released by Apogee and recently acquired by Steinberg, *MasterTools* 1.5 provides a host of tools for polishing your mixes.

Darla just turned 24. Champagne, anyone?

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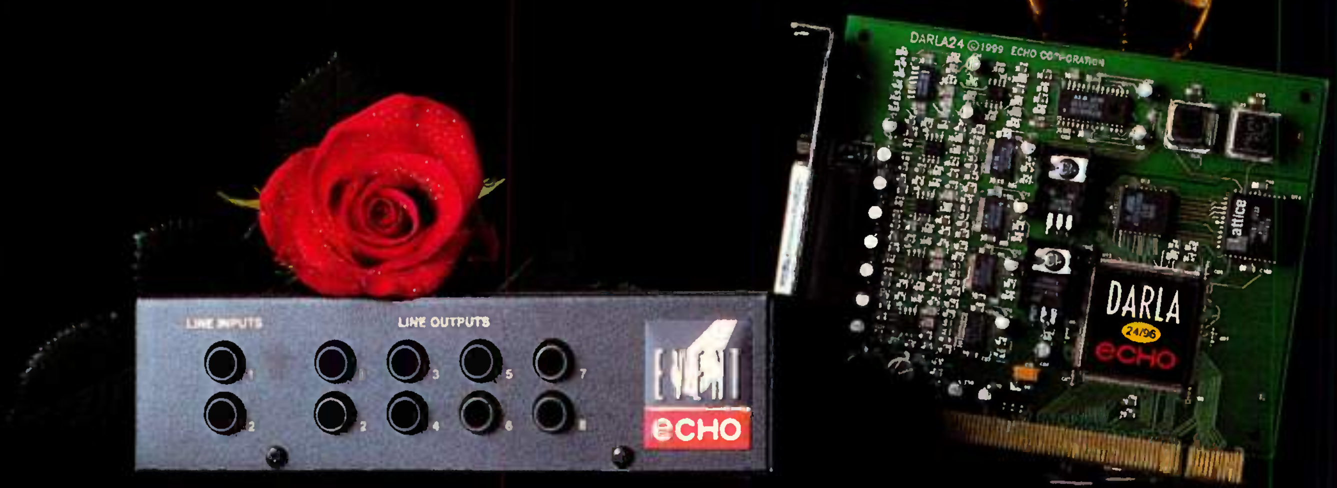

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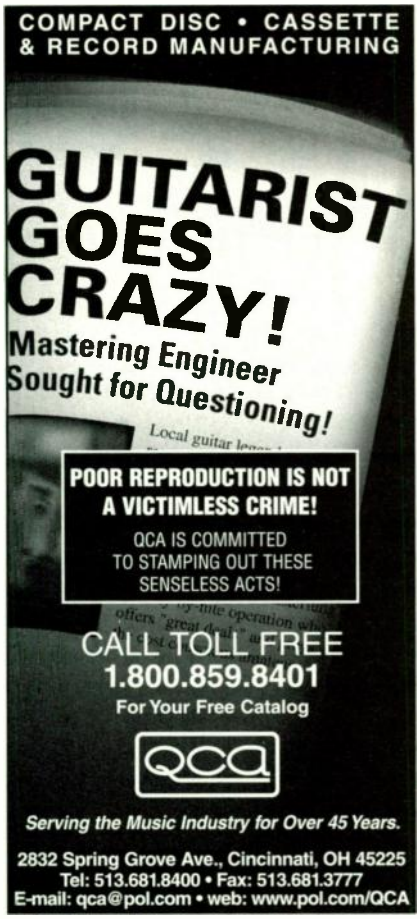

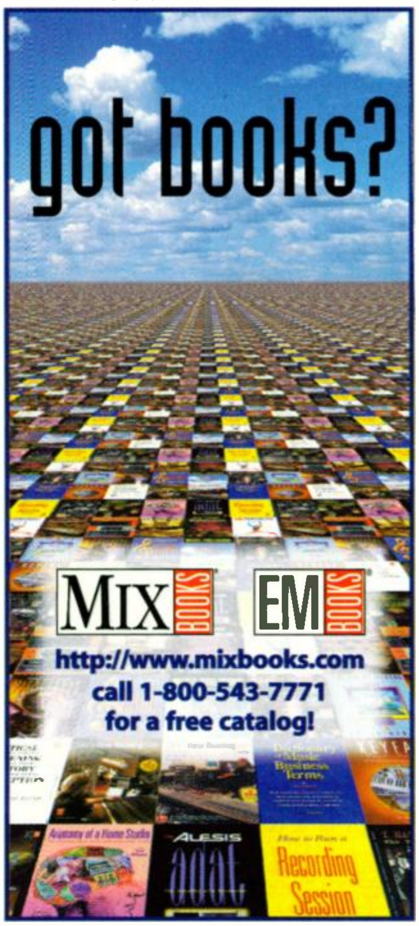

As a result, interest in the program was initially limited.

Happily, those days are over. Steinberg and Spectral Designs have licensed the original code and rebuilt this unique plug-in from the ground up. *MasterTools* 1.5 ($799) is still available for TDM, and later this year it will be released for the VST and DirectX formats, as well.

What's New?

On the outside, not much has changed since Apogee's original *MasterTools* 1.0 release; however, on the inside, there are a lot of cool new features to tell you about. *MasterTools* is now 24-bit compatible, which is great for Pro Tools/24 users. Perhaps the most exciting new feature is the ability to automate all *MasterTools* parameters using *Pro Tools* 4.0 (or newer) software. This allows you to do things that no other TDM plug-in currently does. You can convert a stereo mix into a mono mix during a bridge, swap left and right audio channels on every other beat, or invert the phase of an entire mix for true head-turning effects. This level of functionality makes *MasterTools* a must-have plug-in.

You also get other features, including Nova, an intelligent digital limiter that keeps tabs on any sample approaching a predefined level (also known as an "over") and reduces its level by one least significant bit. This feature lets you rest assured that your mix will not be rejected at the duplicating house. Additionally, the Over List Events (a histogram that keeps track of digital levels over 0 dB) is now shown in frames, where before it could only be displayed in seconds.

Delicious Dithering

MasterTools uses Apogee's proprietary UV22 dithering process. Currently, audio CDs can deal with only 16-bit material, which means that 20- or 24-bit masters must be dithered down to 16 bits before you can print a CD. So how do you maintain the original fidelity? Apogee feels that the UV22 process solves the problem by capturing all of the higher detail and resolution of the source material while still maintaining a warm analog quality, free of the harshness typically associated with 16-bit recordings. Great idea, but does it really work? Based on my tests I'd say yes, it sure does—and at a fraction of the cost of buying Apogee's dedicated hardware to perform the same task.

The Final Call

MasterTools 1.5 includes more improvements than I have room to talk about here (for example, the excellent manual), but I also have a few caveats. Support for older 68 kHz Nubus systems has been dropped, and no direct support is offered for the new Pro Tools/Mix cards. (An upgrade including support for the Mix cards should be available by the time you read this.) Lastly, I noticed that saved presets are not always recalled correctly. Perhaps this is a software glitch. These minor gripes aside, though, I think *MasterTools* 1.5 is a great addition to anyone's audio toolbox.

Overall EM Rating (1 through 5): 4.5

CIRCLE #447 ON READER SERVICE CARD

MUSIC VALVE ELECTRONICS

Vacuum Tube Direct Box

By Peter Freeman

An important link in the audio chain of any studio that is often overlooked is the humble direct injection (DI) box. Its job is to make sure that your low-level instruments make their way safely to the balanced, +4 dBU world of your mic preamp, and ultimately to your mixer or DAW. It's surprising how little thought is given to DI devices much of the time, usually to the detriment of the instruments whose signals have to struggle through cheap and nasty circuitry that compromises their sound rather than preserving or enhancing it. Music Valve Electronics has clearly thought about this state of affairs, and—judging by the quality of its Vacuum Tube Direct Box ($550)—wants to provide a remedy.

Built to Last

The Vacuum Tube Direct Box is a weighty, lavender-colored beast that represents an utterly uncompromising approach to DI design. According to Music Valve's Nathaniel Priest, it is composed of the best components on the market (some of which were custom manufactured to Music Valve's specifications) so that it maintains the highest possible fidelity. These components include a beefy custom-wound toroidal power transformer, a shielded output transformer, metal-film resistors, polypropylene capacitors, and ceramic tube sockets with silver contacts, all of which are expensive and rarely found in devices of this type.

Music Valve professes that the Vacuum Tube Direct Box has a frequency response of 20 Hz–20 kHz (±1 dB) and a noise floor of

-100 dB—claims that I found to be quite credible after using the box. It should be noted that this is a unity-gain device; that is, it causes no gain or loss to your signal.

Particularly notable is that Music Valve made the DI roadworthy. The circuit board and tubes are internally shock-mounted to prevent damage from impact and vibration, major issues while on the road. Also, the simple overall solidity of the unit's enclosure inspires confidence that it will easily withstand the abuse of touring. This is clearly not your average box.

Super Sonics

Sonically, the results here are impressive: this is the best-sounding DI I've used. I normally route line-level signals through the DI section of a high-end microphone preamp/EQ device, which has always sounded very good to me. Somehow, though, the Music Valve DI box gives you an extra musical edge that's hard to describe. It seems that the Vacuum Tube Direct Box not only gets the instrument signal from point A to point B but also enhances it in a subtle way.

I tried it on guitars, basses, and synthesizers and got pleasing results each time (though the box's character was most apparent on the guitars and bass). A/B comparisons with a well-known budget DI were frankly shocking: the sound seemed to suddenly disappear behind a big, woolly blanket, which promptly vanished when I plugged back into the Music Valve box.

Quality Counts

Whether you work in an Alesis ADAT/Mackie–based personal studio or in a Studer/Euphonix–equipped professional recording facility, the quality of your DI box is important. And there are several boxes out there to choose from. I definitely

Music Valve Electronics' new Vacuum Tube Direct Box makes use of high-quality components and is built solidly enough to take on the road.

recommend that you give the Vacuum Tube Direct Box a listen before you buy anything else.

Overall EM Rating (1 through 5): 4.5

CIRCLE #448 ON READER SERVICE CARD

BEATBOY

Richie Gajate-Garcia Authentic Latin Percussion and Drums (Mac/Win)

By Jeff Obee

You're in the middle of a project, the clock is ticking, and you need some burning Latin percussion grooves—alas, no virtuoso player is available to provide them in due time. Beatboy's *Richie Gajate-Garcia Authentic Latin Percussion and Drums* disk ($49.95), from the company's Artist Signature Series, could be the answer. The floppy disk is filled with Standard MIDI Files that lay down some serious Latin rhythms, and it was put together by master percussionist and drummer Richie Gajate-Garcia, who is currently touring with Phil Collins's big band and has performed with Tito Puente, Diana Ross, and Hiroshima. You can also download the collection from Beatboy's Web site, and an audio sample CD is available for $69.95.

The Beats, Boy

The 25 grooves here come in the form of Standard MIDI Files and are presented in GM, XG, and GS formats for automatic mapping. A General MIDI Note Number table is provided to assist you in mapping to non-GM sound sources.

The emphasis here is on percussion, as opposed to a drum kit, and Gajate-Garcia gives you a wide range of standard Latin styles. You'll find salsa, cumbia, bolero, Haitian, songo, Afro-Cuban, merengue, and other styles of sequences, all of which were recorded using Clavia ddrum controllers.

The files average 64 bars long (between 18 and 116), and each contains some combination of intros and outros, verses, choruses, bridges, solos, and fills for plenty of diversity. A two-bar lead-in on each file also contains Program Change data, so the drum sounds in your GM module load automatically.

Using the Beats and Bits

The disk couldn't be easier to use. Open or import the file you want, copy and paste it into the desired location, assign those tracks to the MIDI channel that corresponds with your GM drum/percussion sounds (the default is channel 10), and off you go. Copy an eight-bar section and loop it; cut a small piece of a solo or break to fit into your tune, or use the entire file if you choose.

Appealing Beats

I used the excellent Sweetwater *Total Stereo Session Drums* GM programs for this review. They sounded great, and I couldn't help but pick up my bass and play along.

Everything here grooves solidly and with flair—don't think for a moment that a percussionist playing a MIDI kit detracts from the feel. Gajate-Garcia's playing exhibits the churning, powerful pocket and the polyrhythmic "lilt" that all great Latin percussionists have. The guaguanco, for example, is a hot salsa groove, with the kick

Don't think for a moment that a percussionist playing a MIDI kit detracts from the feel. Beatboy's *Richie Gajate-Garcia Authentic Latin Percussion and Drums* is filled with Standard MIDI Files that lay down some serious Latin rhythms.

bringing in the "one-ee-and-DA, two-ee-AND-da" bass rhythm a third of the way into the file. In this file the congas and bongos hold down the center of the groove while the hi-hat and agogo trade the cascara rhythm throughout; two-thirds of the way in, a cowbell enters and nails the quarter notes. You also get a timbale solo.

Final Beat

At $49.95, the disk is an excellent value—not only for electronic musicians but also as a practice tool for aspiring Latin-style instrumentalists. This is a terrific way to learn your Latin rhythms!

Overall EM Rating (1 through 5): 4

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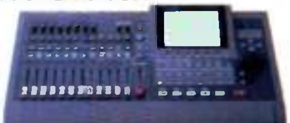
HARD DISK RECORDERS

Roland

VS1680 Digital Production Studio



The VS-1680 Digital Studio Workstation is a complete 16 track, 24-bit recording, editing, mixing and effects processing system in a compact tabletop workstation. The latest sytem upgrade for the VS1680 includes Cosm speaker modelling and Master Toolkit effects and up to 18 tracks of recording and playback.

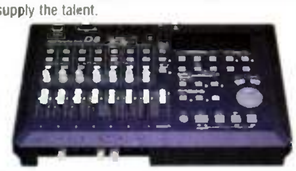
FEATURES–

- 16 tracks of hard disk recording, 256 virtual tracks
- 24-bit MT Pro Recording Mode for massive headroom and dynamic range
- Large 320 x 240 dot graphic LCD provides simultaneous level meters, playlist, EQ curves, EFX settings, waveforms and more
- 20-bit A/D D/A converters
- 2 optional 24-bit stereo effects processors (VS8F-2) provide up to 8 channels of independent effects processing
- 12 audio outs: 8x RCA, 2x stereo digital & phones
- New EZ routing function allows users to create and save various recording, mixing, track bouncing, and other comprehensive mixer templates for instant recall
- 10 audio inputs: 2 balanced XLR-type inputs w/ phantom power, 6 balanced 1/4" inputs, and 1 stereo digital input (optical/coaxial)
- Direct audio CD recording and data backup using optional VS-CDR CD recorder

KORG

D8 Digital Recording Studio

The new D8 Digital Recording Studio features an 8-track recorder, a 12-channel mixer, onboard effects, and basically everything else you'll need to record and mix your music, you supply the talent.



FEATURES–

- 8-track recorder, 12-channel mixer
- 1.4GB hard disk for up to 4.5 hours of recording on a single track
- High and low EQ on each channel
- 130 high-quality stereo digital effects for complete recording in the digital domain
- MIDI clock sync, SCSI port and S/PDIF digital interfaces all standard

FOSTEX

FD8 8-track Hard Disk Recorder

8 tracks of Digital Audio from one of the leaders in the industry, at a killer price! Records to a variety of SCSI compatible drives.

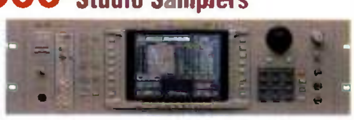
FEATURES–

- ADAT Lightpipe I/O for exchanging up to 8-tracks directly to ADAT
- Random access editing features include Cut/Copy/Paste/Move plus undo and redo
- Uncompressed 44.1kHz, 16-bit sampling
- Dual XLR mic inputs complete with trim on channels 7 and 8 for low-Z mics
- S/PDIF Optical I/O• 8 x 2 mixer with 3-band EQ

DIGITAL SAMPLERS

AKAI

S5000 & S6000 Studio Samplers



Akai is proud to announce its next generation of samplers with the introduction of the S6000 and the S5000. Building upon Akai's legendary strengths, both machines feature up-to 128-voice polyphony and up-to 256 MB of RAM. They use the DOS disk format and .WAV files as the native sample format allowing standard PC .WAV files to be loaded directly for instant playback - even samples downloaded from the Internet into your PC may be used. And of course, both the S6000 and S5000 will read sounds from the S3000 library.

FEATURES–

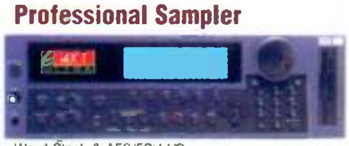
- OS runs on easily upgradeable flash ROM
- 2x MIDI In/Out/Thru ports for 32 MIDI channels
- Stereo digital I/O and up to 16 analog outputs
- 2x SCSI ports standard • Wordclock connection
- Optional ADAT interface provides 16 digital outs
- .WAV files as native sample format

S6000 ONLY FEATURES–

- Removable front panel display
- User Keys
- Audio inputs on both the front and rear panel allow you to wire the S6000 directly into a patchbay from the back and override this connection simply by plugging into the front

E-MU E-mu Systems, Inc.

E4XT ULTRA Professional Sampler



The Emulator legacy continues with the new ULTRA series from E-mu. Based on the EIV samplers the new 32-bit RISC processing of the E4XT guarantees faster MIDI response, SCSI, DSP and sampling.

FEATURES–

- 128 voice polyphony
- 64mb RAM (exp. to 128)
- 3.2GB Hard Drive • Dual MIDI (32 channels)
- 24-bit effects processor • 8 bal. outs (exp to 16)
- Word Clock & AES/EBU I/O
- EOS 4.0 software • 9 CD ROMS over 2GB snds
- Optional Adat card offers 8 ins/ 16 outs

CD RECORDERS & BURNERS

HHB CDR-850 CD Recorder

The new HHB CDR850 is one of the most comprehensive CD-R, CD-RW recorders available today. It delivers the outstanding sound quality that HHB is known at a lower price than previous models. Equipped with a complete range of analog and digital I/O and easy to use one touch recording modes make the CDR850 suitable for any audio environment no matter how sophisticated or demanding.

- CD-R, CD-RW compatible
- All functions accessible from front panel menu
- 4 one touch recording modes; 2 manual, 2 automatic
- Sample rate converter accepts any digital signal from 32kHz to 48kHz including varispeed
- Copies all CD, DAT, MD, DVD and DCC tarck starts
- Complete user control over SCMS
- Balanced XLR analog I/O, Unbalanced (RCA) phono analog I/O, AES/EBU digital input, coaxial & optical S/PDIF digital I/O

MICROBOARDS

CopyWriter A2D CD Duplication System

The first CD to CD standalone duplicator with built-in Analog to Digital Conversion capability. Easy to use and powerful, the A2D has a 2.1GB internal hard drive and a SCSI port for direct connection to a Mac or PC. A perfect solution for audio, data and video applications.

Features–

- Interface includes Microphone in, Audio line in, Audio line out and external SCSI port
- Supported Formats: CD DA, CD ROM mode 1 & 2, XA, CD Bridge, Photo CD, CD Extra, Multi Session, Mixed Mode, Karaoke, (optional)
- Duplication Speed: 8X Read/ 4X Write
- Windows 95, NT, 3.1, Mac OS and Unix compatible
- Headphone output with level control

Jam Session PlayWrite 4080

The Jam Session PlayWrite 4080 from Microboards is an all-in-one SCSI CD recorder specifically packaged for audio CD pre-mastering on the Mac OS platform. Built around a 4X write 8X read Matsushita CD-R mechanism the Jam Session comes bundled with all of the pro level software nessecary to edit, master and sequence audio files for CD burning.

Features–

- 4X write/8X read Matsushita CD-R mechanism
- Includes Red Book compliant Adaptec Jam software for editing PQ codes and crossfades for audio CD pre-mastering, will also break down Sound Designer II regions into seperate audio files for editing
- Includes Bias Peak Le for recording/editing and optimizing audio files before burning
- Also includes Adaptec Toast for Data backup, CD Rom CD I, and CD Video authoring
- Includes 2 Microboards CD-R's and SCSI cable
- Requires Power PC running Mac OS

STUDIO DAT-RECORDERS

Panasonic SV-3800

The SV-3800 features a highly accurate and reliable transport mechanism with search speeds of up to 400X normal. It use 20-bit D/A converters to satisfy even the highest professional expectations. Panasonic DATs are found in studios throughout the world and are widely recognized as the most reliable DAT machines available on the market today.

FEATURES–

- 64x Oversampling A/D converter for outstanding phase characteristics
- Search by start ID or program number
- Single program play, handy for post.
- Adjustable analog input attenuation, +4/-10dBu
- L/R independent record levels
- Front panel hour meter display
- 8-pin parallel remote terminal
- 250x normal speed search

TASCAM DA-40

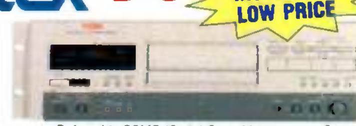
Upholding the standards of the world renowned DA-30 series DAT machines for durability and sonic excellence, the DA-40 adds some advanced features such as track names and digital input format sensing.

FEATURES–

- XLR balanced & RCA (phono) unbalanced analog I/O
- AES/EBU and S/PDIF digital I/O
- Play and record @ 32, 44.1 and 48kHz sample rates
- Jog/Shuttle wheel
- Alphanumeric data entry for naming programs
- Independent input level adjustment capability
- Output trim for XLR balanced outputs
- Selectable SCMS code function
- **Optional RC-D45 Remote Controller**

FOSTEX D-5

INCREDIBLE LOW PRICE



The Fostex D-5 is a full featured yet suprisingly affordable professional DAT machine. Balanced XLR I/O and AES/EBU are two of the unique features for a digital audio tape recorder at this price point.

FEATURES–

- 1-bit analog to digital and digital to analog converters
- Balanced XLR analog I/O switchable between +4 and -10 dBu • AES/EBU and S/PDIF optical digital I/O
- Standard and Long Play mode record and playback
- Defeatable SCMS (Serial Copy Management System)
- 48, 44.1 and 32kHZ sample rates are supported
- Jog/Shuttle capabilities

INCLUDES: Infra red remote control

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DIGITAL MIXERS

Panasonic

WR-DA7 Digital Mixing Console

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

You want to see what all the digital mixing buzz is about? The NEW TMD1000 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.

DIGITAL MULTI-TRACK RECORDERS

TASCAM

DA-88 Modular Digital Multitrack

The standard digital multitrack for post-production and winner of the Emmy award for technical excellence, the DA-88 delivers the best of Tascam's Hi-8 digital format. Its Shuttle/Jog wheel and track delay function allow for precise cueing and synchronization and the modular design allows for easy servicing and performance enhancements with third-party options.

FEATURES-
- 1 48 minutes record time on a single 120 min tape
- Expandable up to 128 Tracks using 16 machines
- User-definable track delay & crossfade
- Shuttle & Jog capability
- Auto punch with rehearsal
- SMPTE, MIDI and Sony 9-Pin sync capability
- Options include RC-828/898 Remote Controllers, IF-AEB/IF-88SD digital interfaces, MU-Series meter bridge, MMC-88 MIDI machine control interface, SY-88 Sync Card

DA-38 Digital Multitrack for Musicians

Designed especially for musicians, the DA-38 is an 8 track digital recorder that puts performance at an affordable price. It features an extremely fast transport, Hi-8 compatibility, rugged construction, ergonomic design and sync compatibility with DA-88s.

ALESIS

ADAT XT20 Digital Audio Recorder

The New ADAT-XT20 provides a new standard in audio quality for affordable professional recorders while remaining completely compatible with over 100,000 ADATs in use worldwide. The XT20 uses the latest ultra-high fidelity 20-bit oversampling digital converters for sonic excellence, it could change the world.

FEATURES-
- 10-point autolocate system
- Dynamic Braking software lets the transport quickly wind to locate points while gently treating the tape.
- Remote control
- Servo-balanced 56-pin ELCO connector
- Built-in electronic patchbay
- Copy/paste digital edits between machines.

ADAT LX20 Digital Audio Recorder

The most affordable ADAT ever made, the new LX20 features true 20-bit recording at a price you won't believe. Compatibility with all other ADATs and digital consoles, the LX20 provides the same sync options and digital inputs as the big brother XT20 at a lower price point.

ADAT OPTIONS-
- **BRC** for all Adat (except M20) w/ 460 locate pts, smpte/absolute time & bar and beat timing references, digital editing and transport control for up to 16 ADATs
- **AI3** 20-bit 8 channel analog - optical I/O interface
- **CADI** remote control/autolocator for M20 w/ jog/shuttle & rj-45 ethernet connector for long distance cable runs
- **Adat/Edit** integrated PCI digital audio card and software for recording and editing on Mac & Windows computers

MIC PREAMPS

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't 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 586 Vacuum Tube Mic Pre

Hand selected and matched premium 12AU7 vacuum tubes ensure ideal characteristics for a warm, distortion free signal path. Versatile enough to use with virtually any input source.

FEATURES-
- Mic or line/instrument inputs on each channel. +4/-10 operation.
- Drive control for a wide variety of great tube effects
- Custom Analog Vu Metering
- 3-Band EQ with sweepable frequency
- Optional TYPE IV Conversion System outputs
- Separate 1/4" insert send/return on each channel

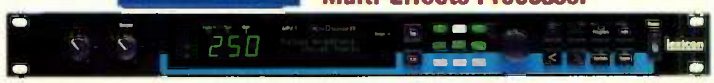
EFFECTS PROCESSING

t.c. electronic M3000 Professional Reverb

Incorporating TC Electronic's new VSS-3 technology, the M3000 is a great sounding, versatile reverb that is easy to use. Combining ultimate control of early reflections with a transparent reverb tail, the art of reverberation is brought to a new level. Whether it's a phone booth, cave or concert hall, the M3000 delivers high-quality ambience.

FEATURES-
- VSS-3, VSS-3 Gate, C.O.R.E. & REV-3 reverbs as well as Delay, Pitch, EQ, Chorus, Flanger, Tremolo, Phaser, Expander/Gate, Compressor and De-Esser
- 300 high-grade factory presets including Halls, Rooms, Plates, Ambience, Gated Reverbs, and more
- Up to 300 user presets in internal RAM and 300 more using an optional PCMCIA card.
- Dual engine configuration featuring 24-bit A/D/D/As.
- Connections include AES/EBU, Coaxial S/PDIF, Optical Tos-Link/ADAT & analog XLR I/Os, MIDI IN/OUT/THRU, Clock Sync and External Control.

lexicon MPX1 Multi-Effects Processor



The MPX-1 is truly an outstanding multi-effects device. Using Lexicon's Lexchip, it offers outstanding reverb or ambience as well as a separate processor for effects for awesome power in the studio or on the road.

FEATURES-
- Intuitive user interface for easy editing, built-in help.
- Balanced Analog I/O (1/4" & XLR)
- 56 effect algorithms
- Digital Inputs & Outputs (S/PDIF @ 44.1KHz)
- 18 Bit A/D; 20 Bit D/A Conversion, 32-bit processing
- >90dB of Dynamic Range
- Intelligent Sorting by Name, Number, Application, etc.
- Parameter Morphing
- Dynamic MIDI® patching & MIDI automation

COMPRESSORS

PreSonus ACP88 8 Channel Compressor

The ACP88 comprises eight channels of compression, limiting and noise gating for a variety of studio applications. It features individual side chain for each channel and it's attractive blue anodized finished lets you show your true sonic colors.

FEATURES-
- 8 separate compressors/gates with individual controls. • Servo balanced or unbalanced inputs & floating balanced or unbalanced outputs. • Individual side chain jacks for spectral compression and a separate sidechain jack for gate processing. • Each channel boasts full gain reduction metering, compression threshold indication & gate open/close. • Front panel buttons include hard/soft knee compression, peak/auto compression, bypass, gate range and link.
- Link feature uses a unique summing bus for multiple combinations of master/slave link setups.

dbx QUANTUM 24-bit Mastering Processor

FEATURES-
- 96kHZ/24-bit A/D-D/A 48-bit internal • 4-band compressor, gate, limiter • 5-band EQ w/ Hi & Lo shelving & 3 fully parametric bands • Normalize, Stereo width adjust • Dither • Sample Rate Conversion • 4 band crossover w/ variable slopes • proprietary sync chips for extremely low jitter • T.S.E. tape saturation emulation • Adds warmth, body and punch to your mix

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The new D8 Digital Recording Studio features an 8-track recorder, a 12-channel mixer, onboard effects, and basically everything else you'll need to record and mix your music, you supply the talent.

FEATURES–

- 8-track recorder, 12-channel mixer.
- 1.4GB hard disk for up to 4.5 hours of recording on a single track.
- High and low EQ on each channel.
- 130 high-quality stereo digital effects for complete recording in the digital domain.
- MIDI clock sync, SCSI port and S/PDIF digital interfaces all standard

Fostex

FD8 8-track Hard Disk Recorder

8 tracks of Digital Audio from one of the leaders in the industry, at a killer price! Records to a variety of SCSI compatible drives.

FEATURES–

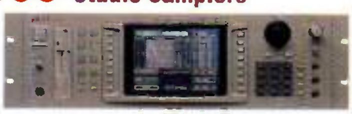
- ADAT Lightpipe I/O for exchanging up to 8-tracks directly to ADAT
- Random access editing features include Cut/Copy/Paste/Move plus undo and redo
- Uncompressed 44.1kHz, 16-bit sampling
- Dual XLR mic inputs complete with trim on channels 7 and 8 for low-Z mics
- S/PDIF Optical I/O• 8 x 2 mixer with 3-band EQ.

DIGITAL SAMPLERS

AKAI

S5000 & S6000 Studio Samplers

Akai is proud to announce its next generation of samplers with the introduction of the S6000 and the S5000. Building upon Akai's legendary strengths, both machines feature up-to 128-voice polyphony and up-to 256 MB of RAM. They use the DOS disk format and .WAV files as the native sample format allowing standard PC .WAV files to be loaded directly for instant playback - even samples downloaded from the Internet into your PC may be used.And of course, both the S6000 and S5000 will read sounds from the S3000 library.



FEATURES–

- OS runs on easily upgradeable flash ROM.
- 2x MIDI In/Out/Thru ports for 32 MIDI channels
- Stereo digital I/O and up to 16 analog outputs.
- 2x SCSI ports standard • Wordclock connection
- Optional ADAT interface provides 16 digital outs
- .WAV files as native sample format

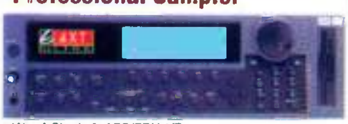
S6000 ONLY FEATURES–

- Removable front panel display
- User Keys
- Audio inputs on both the front and rear panel allow you to wire the S6000 directly into a patchbay from the back and override this connection simply by plugging into the front.

E-MU E-mu Systems, Inc.

E4XT ULTRA Professional Sampler

The Emulator legacy continues with the new ULTRA series from E-mu. Based on the EIV samplers the new 32-bit RISC processing of the E4XT quarantees faster MIDI response, SCSI, DSP and sampling.



FEATURES–

- 128 voice polyphony
- 64mb RAM (exp. to 128)
- 3.2GB Hard Drive • Dual MIDI (32 channels)
- 24-bit effects processor • 8 bal. outs (exp to 16)
- Word Clock & AES/EBU I/O
- EOS 4.0 software • 9 CD ROMS over 2GB snds
- Optional Adat card offers 8 ins/ 16 outs

CD RECORDERS & BURNERS

HHb CDR-850 CD Recorder

The new HHB CDR850 is one of the most comprehensive CD-R, CD-RW recorders available today. It delivers the outstanding sound quality that HHB is known at a lower price than previous models. Equipped with a complete range of analog and digital I/O and easy to use one touch recording modes make the CDR850 suitable for any audio environment no matter how sophisticated or demanding.

- CD-R, CD-RW compatible
- All functions accessible from front panel menu
- 4 one touch recording modes; 2 manual, 2 automatic
- Sample rate converter accepts any digital signal from 32kHz to 48kHz including varispeed
- Copies all CD, DAT, MD, DVD and DCC tarck starts
- Complete user control over SCMS
- Balanced XLR analog I/O, Unbalanced (RCA) phono analog I/O, AES/EBU digital input, coaxial & optical S/PDIF digital I/O

MICROBOARDS

CopyWriter A2D CD Duplication System

The first CD to CD standalone duplicator with built-in Analog to Digital Conversion capability. Easy to use and powerful, the A2D has a 2.1GB internal hard drive and a SCSI port for direct connection to a Mac or PC. A perfect solution for audio, data and video applications.

Features–

- Interface includes Microphone in, Audio line in, Audio line out and external SCSI port
- Supported Formats: CD DA, CD ROM mode 1 & 2, XA, CD Bridge, Photo CD, CD Extra, Multi Session, Mixed Mode, Karaoke, (optional)
- Duplication Speed: 8X Read/ 4X Write
- Windows 95, NT, 3.1, Mac OS and Unix compatible
- Headphone output with level control

Jam Session PlayWrite 4080

The Jam Session PlayWrite 4080 from Microboards is an all-in-one SCSI CD recorder specifically packaged for audio CD pre-mastering on the Mac OS platform. Built around a 4X write 8X read Matsushita CD-R mechanism the Jam Session comes bundled with all of the pro level software nessecary to edit, master and sequence audio files for CD burning.

Features–

- 4X write/8X read Matsushita CD-R mechanism
- Includes Red Book compliant Adaptec Jam software for editing PQ codes and crossfades for audio CD pre-mastering, will also break down Sound Designer II regions into seperate audio files for editing
- Includes Bias Peak Le for recording/editing and optimizing audio files before burning
- Also includes Adaptec Toast for Data backup, CD Rom CD I, and CD Video authoring
- Includes 2 Microboards CD-R's and SCSI cable
- Requires Power PC running Mac OS

STUDIO DAT-RECORDERS

Panasonic SV-3800

The SV-3800 features a highly accurate and reliable transport mechanism with search speeds of up to 400X normal. It use 20-bit D/A converters to satisfy even the highest professional expectations. Panasonic DATs are found in studios throughout the world and are widely recognized as the most reliable DAT machines available on the market today.

FEATURES–

- 64x Oversampling A/D converter for outstanding phase characteristics
- Search by start ID or program number
- Single program play, handy for post.
- Adjustable analog input attenuation, +4/-10dBu
- L/R independent record levels
- Front panel hour meter display
- 8-pin parallel remote terminal
- 250x normal speed search

TASCAM DA-40

Upholding the standards of the world renowned DA-30 series DAT machines for durability and sonic excellence, the DA-40 adds some advanced features such as track names and digital input format sensing.

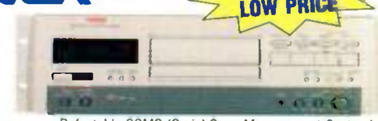
FEATURES–

- XLR balanced & RCA (phono) unbalanced analog I/O
- AES/EBU and S/PDIF digital I/O
- Play and record @ 32, 44.1 and 48kHz sample rates
- Jog/Shuttle wheel
- Alphanumeric data entry for naming programs
- Independent input level adjustment capability
- Output trim for XLR balanced outputs
- Selectable SCMS code function
- **Optional RC-D45 Remote Controller**

Fostex D-5

INCREDIBLE LOW PRICE

The Fostex D-5 is a full featured yet suprisingly affordable professional DAT machine. Balanced XLR I/O and AES/EBU are two of the unique features for a digital audio tape recorder at this price point.



FEATURES–

- 1-bit analog to digital and digital to analog converters
- Balanced XLR analog I/O switchable between +4 and -10 dBu • AES/EBU and S/PDIF optical digital I/O
- Standard and Long Play mode record and playback
- Defeatable SCMS (Serial Copy Management System)
- 48, 44.1 and 32kHZ sample rates are supported
- Jog/Shuttle capabilities

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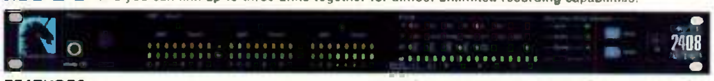
COMPUTER BASED DIGITAL AUDIO SYSTEMS

The recent proliferation of computer based digital audio worksatations (DAWs) is enough to make even the most seasoned audio professional's head spin. Is it com patible with my software? How will it interface with my current gear? Does it have the I/O I need. How about expandability? B&H has the answers. We have a wide selection of the most popular digital audio cards and systems available to fit your budget and needs no matter how big or small.

MOTU 2408 Hard Disk Recording System

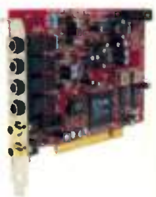
The new Mark of the Unicorn 2408 is turning the industry upside down! No other system in it's price range gives you performance like this, with a full simultaneous 24 inputs and outputs on a custom designed VLSI chip that is dedicated to quality I/O. The 2408 is 24-bit compatible and you can link up to three units together for almost unlimited recording capabilities.



FEATURES–
- 7 banks of 8 channel I/O. 1 bank of analog, 3 banks of ADAT optical, 3 banks of Tascam TDIF, plus stereo S/PDIF.
- Custom VLSI chip for amazing I/O capabilities
- Connect up to three 2408 units to your computer for a total of 72 input and output connections.
- Format conversion between ADAT and DA-88
- 20-bit A/D and D/A converters on analog ins & outs
- 24-bit internal data bus for full 24-bit recording via digi tal inputs
- Standard S/PDIF I/O for digital plus an additional S/PDIF I/O for the main mix
- Sample-accurate synchronization with ADATs and DA88s via an ADAT SYNC IN and RS422
- Includes a complete waveform editing program for Power Macintosh
- Will grow as your computer grows

SPECIAL! 2408 system with Digital Performer competitive upgrade for $1199.95

Digital AUDIO Card Deluxe
24-bit/96kHz Audio Card

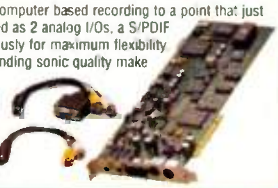


From the company that's been bringing sonic excellence to the Windows platform for nearly a decade comes the affordable Card Deluxe. It's a half length, no compromise, 24-bit/96kHz PCI card compatible with todays cutting edge production software. You can even chain multiple cards together for multiple sample accurate I/O's. Available now for Windows with support for the MacOS coming soon.

FEATURES–
- 8 to 24-bit resolution
- 22 to 96kHz sampling rate
- 2 channel 1/4" TRS balanced analog I/O
- Coaxial S/PDIF digital I/O • Full duplex
- +4/-10 balanced/unbalanced operation
- 4 channel operation using both analog and digital I/O
- Slave multiple Card Deluxes to single sample clock using DAL's WavSync drivers
- Windows 95, 98 and NT drivers
- DirectX support

KORG 12/12 I/O
Multi-channel PCI Audio Card

The 1212I/O card helps bring the price of full function multi-channel computer based recording to a point that just about anyone can afford. It features 12 inputs and outputs configured as 2 analog I/Os, a S/PDIF I/C and and 8-channel ADAT optical I/O. All I/Os can be used simultaneously for maximum flexibility. Compatiblity with most Digital Audio Software on the market and outstanding sonic quality make the 1212I/O a great choice for project studios and multimedia pros.



FEATURES–
- Total of 12 ins & outs, all can be used simultaneously
- 44.1 and 48kHz sample rates
- 20-bit enhanced dual bit Inputs, 18-bit outputs.
- 20-20kHz frequency response
- Compatible with any PCI Macintosh or Windows computer

Lexicon Studio Recording System

The Lexicon Studio System interfaces with your favorite digital audio software for a complete hard disk recording package. Supporting both PC and Mac, Lexicon Studio can be expanded up to 32 voices from a variety of I/O options. For recording, editing, mixing and DSP, Lexicon Studio is here.

FEATURES–
- The Core-32 System PCI-Card is capable of supporting 32 audio streams simultaneously. It can also be used as a time code or clock master or slave.
- The PC-90 Digital Reverb daughterboard attaches to the Core-32 providing 2 discrete stereo reverbs
- The LDI-12T delivers up to 12 channels of simultaneous I/O supporting analog (+4 XLR and -10 RCA), s/pdif, and ADAT.
- Direct support of Steinberg Cubase VST and many other software programs.
- Optional LDI-10T 24-bit audio interface has 8 balanced Ins & outs using 1/4" TRS connectors a coaxial S/PDIF in and out as well as a 1/4" time code input

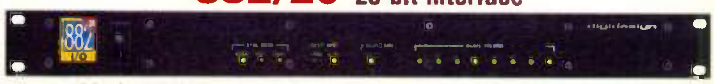
digidesign Project II Bundle

A Division of Avid Technology, Inc.

The new Project II Bundle incorporates Digidesign's legendary SampleCell II card w/32MB RAM, MasterList CD and Logic Audio AV software together with the popular Project II recording card for a price that you are not going to believe! Just add your choice of interface and you have a complete PCI studio at your fingertips.

FEATURES–
- Direct I/O for direct communication between your digital audio sequencer your Digidesign Audio Interface, without the intervention of the Macintosh Sound Manager
- 2 CD ROMS of ready to use sounds
- Complete mastering to Red book standards

882/20 20-bit interface



The new Digidesign 882I20 I/O is a high performance entry-level audio interface for Pro Tools. It features 20-bit D/A & A/D converters, 24-bit digital performance, and an extremely low noise floor. The 882I20 makes an excellent auxiliary audio interface for Pro Tools24 — ideal for connecting outboard signal processing gear, keyboards, or other external devices. It can even operate as a standalone, 2-channel, 20-bit A/D converter or D/A converter without Pro Tools (or any other) software.

ToolBox PCI Digital Audio Bundle For Mac or PC

When you need professional features at an affordable price, the Digidesign ToolBox delivers a great combination of software and hardware for Mac or PC. Based around Digidesign's AudioMedia III a 16-bit audio card with stereo RCA inputs, 1 bit 128x over sampling A/D and 18-bit D/A converters as well as coaxial S/PDIF digital I/O. This system is ideal for personal and project studios, radio broadcast applications, and multimedia audio production.

ToolBox For Mac Includes:–
- Includes **Audiomedia III** card
- **Pro Tools 4.x** recording/editing software with playback support for up to 8 tracks of audio
- **D-fx** AudioSuite Plug-Ins (Reverb, Delay & Modulation Effects)
- **D-fi** AudioSuite sound degeneration Plug-Ins
- **Bias Peak Le** 2 track editing software

ToolBox For PC Includes:–
- Includes **Audiomedia III** card
- **Session** Software recording/editing w/ support for playback of 8 tracks of audio • **Logic Audio AV** MIDI sequencing/ audio software
- **Sound Forge XP** 2 track editing software • **ACID Rock** loop based audio sequencer allowing you to dictate the pitch and tempo of any .wav file

SOFTWARE

MOTU Digital Performer 2.6
MIDI/AUDIO Software for Mac

Their second major update this year, with a relentless stream of new advanced features, like sample-accurate editing, sample-accurate sync and MOTU's innovative RAM-based loop recording tool called POLAR. DP is packed full of featues you won't find anywhere else:

FEATURES-
- Includes over 50 real-time MIDI and audio effects plug-ins • POLAR window - Interactive audio loop recording the way it should be • 24-bit recording and editing • 32-bit native effects processing - incredible sounding EQ and other FX • 64-bit MasterWorks™ Limiter and Multiband Compressor plug-ins included • Advanced waveform editor • Sample-accurate - the most reliable editing and tightest sync you can get • OMF export - transfer your entire session, crossfades and all, into Pro Tools • Samplers window - drag & drop samples between your Mac and your Sampler • PureDSP™ stereo pitch-shifting and time-stretching • Unlimited audio tracks, real time editing, full automation and remote control • QuickTime digital video support, and much more • Compatible with Pro ToolsI24, the MOTU 2408 and today's other popular systems • Digital Performer is an entire recording studio inside your computer

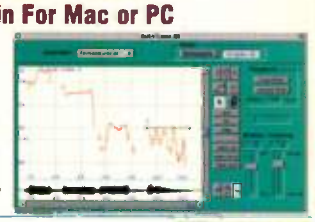
cakewalk Pro Audio 8
MIDI/Audio Software for PC

One of the industries leading MIDI/Audio software with a wide range of hardware support and realtime automation and editing for Windows 95/98 and NT 4.0.

Features–
- 24-bit/96kHz compatible
- 128 audio tracks, 256 Audio/MIDI tracks
- 256 realtime effects w/ 32-bit processing
- Vector based volume/Pan automation
- MIDI effects- quantize, delay/echo, transpose, arpeggiator
- 24 stave notation w/lyrics, chord symbols, guitar chord diagrams and percussion notaion
- SMPTE, MTC and MMC support
- Playback of .AVI, QT and MPEG video

ANTARES Auto Tune
Plug-In For Mac or PC

Intonation correcting multi-platform plug-in for Mac and PC considered to be the "Holy Grail of recording" by Recording magazine. Auto-Tune corrects pitch and intonation problems on voice and solo instruments without distortion or artifacts. Two modes of operation include Automatic where pitch is continously compared to a user selected scale and Graphical mode offering more precise control allows you to draw specific target pitches. Compatible with TDM, VST MAS and standalone on the Mac and DirectX or DAL V8 on the PC.



WAVES Native Power Pack

Uses the CPU of your Mac or PC to provide top quality effects processing for recording, mixing and multi-media applications. Compatible with many popular audio editing software programs, the NPP provides EQ, Reverb, Compression, Gating, Stereo Imaging and the incredible L1 Ultramaximizer mastering peak limiter. It also includes Wave Convert, a stand alone application that batch converts formats, bit-depths & sample rates for the loudest, cleanest multimedia files available. A must have for recording engineers & internet designers alike

Native Power Pack II

The all new Native Power Pack II is an entirely different plug-in collection than the original Native Power Pack. Bass-enhancement, de-essing, vintage compression/expansion and EQ are all provided and can be used with or without the original NPP. You can also upgrade from either NPP or NPPII to the Native Gold bundle for the complete Waves experience, and like the earlier NPP, the NPP II *requires no extra DSP!*

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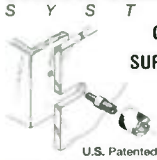

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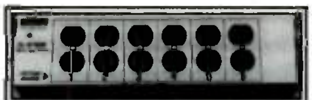

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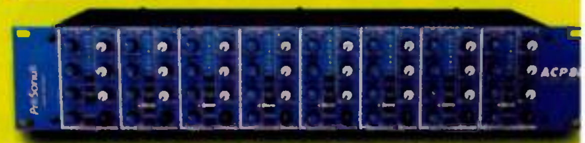
ACP88



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ACP22

Electronic Musician 1999 EDITORS CHOICE

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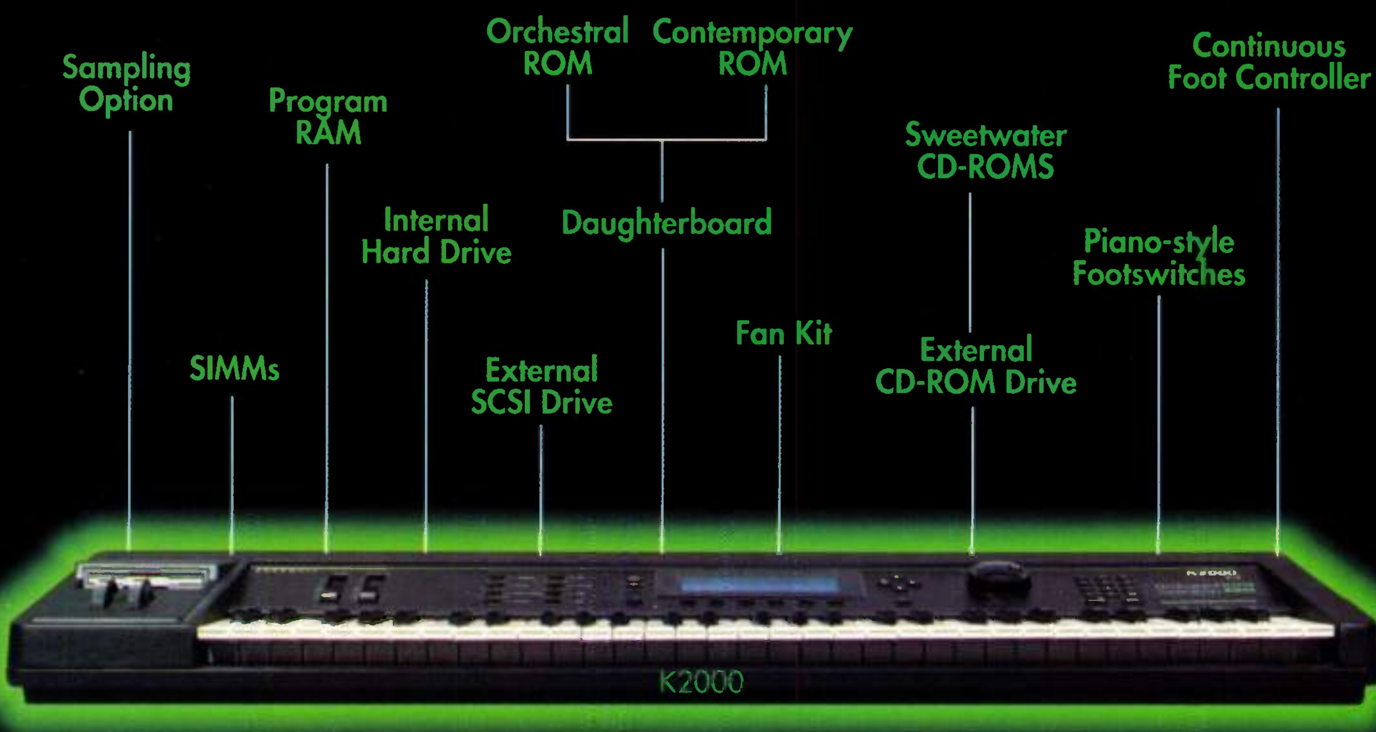

Your music is unique. So why compromise your sound with an "off-the-shelf" synthesizer/sampler? Instead of accepting the ordinary, consider the **Kurzweil K2000** and **K2500 Series** instruments. Yes, it is true that right out of the box, your new Kurzweil keyboard or rack module is already the most powerful musical instrument ever made. But that's just the beginning, because you can custom configure your own Kurzweil for unprecedented expressive capabilities.

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Exactly which options are right for *your* custom Kurzweil? That's were we come in. **Sweetwater Sound** has more experience building and supporting customized Kurzweil instruments than anyone else in the world. In fact, the very first music technology product we ever represented was the legendary K250 system from Kurzweil. Since then we've been building complete systems, creating lush, critically-acclaimed soundware libraries on CD-ROM and helping thousands and thousands of musicians get the most out of their Kurzweils.

Ready for an instrument that transcends the ordinary? Ready for an instrument that's built to your specifications for your music? Ready for the sonic thrill ride of your life? Then you're ready for a custom-configured Kurzweil from Sweetwater.

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- **Live Mode:** External digital or analog signals can now be processed and controlled via Kurzweil's Variable Architecture Synthesis (V.A.S.T.) engine. *(Available for K2500 only)*
- **KB3 Mode:** Realtime Drawbars with all the features of a real B-3 Organ. *(K2500 only)*
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- **Orchestral ROM:** 8 MB of superb quality solo and ensemble Symphonic Instruments.
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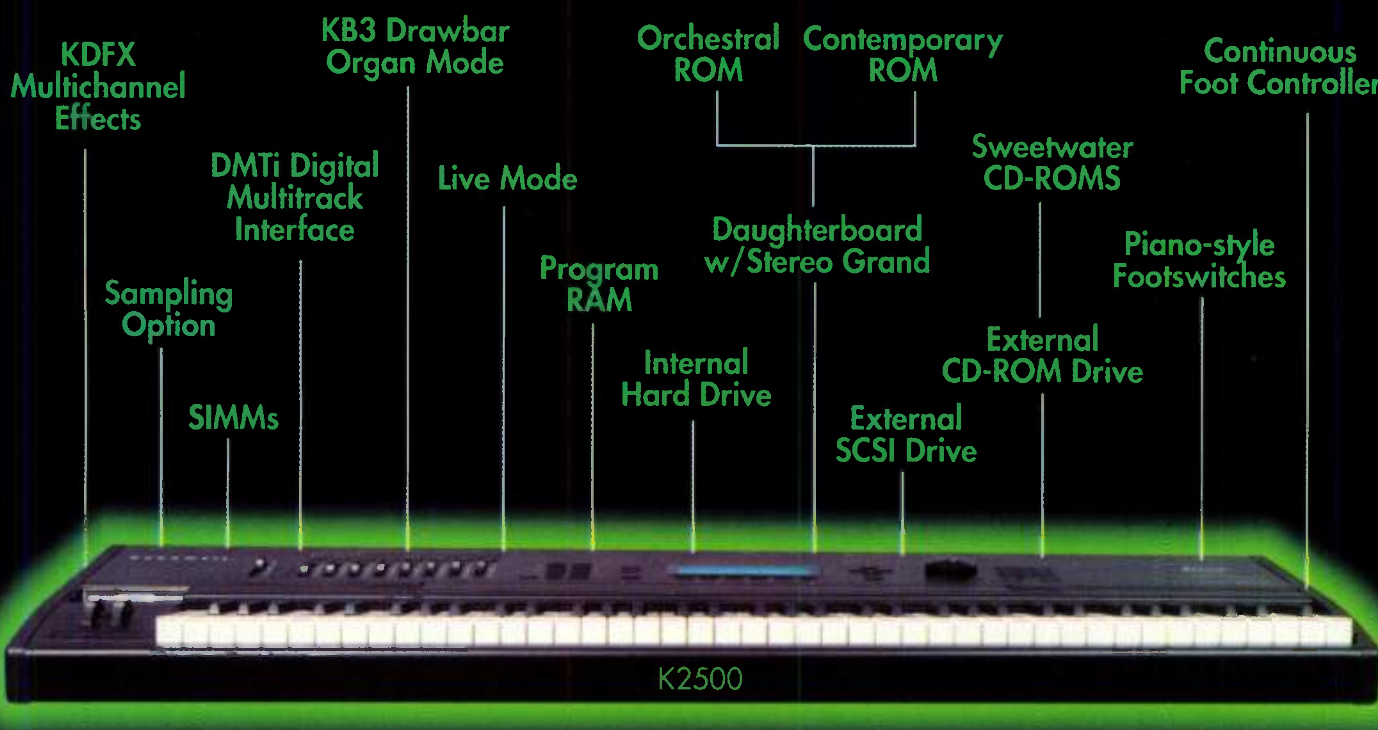

- **Program RAM:** Holds additional Programs and Songs (not Samples) even after power off.
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- **External Hard or Removable Drives:** Virtually limitless storage for all your data.
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FINAL MIX

By Larry the O

Information, Please

No matter how well you master the fine art of learning, the difficulty associated with using a new tool is directly proportional to the amount of information you have about it. Information can be gathered from many places, but the first source that we all turn to is the manufacturer. Some manufacturers provide excellent guidance to help users get started with their products. Two recent examples include the mic-technique summary in the Alesis GT microphones manuals, and the installation information that accompanies Mark of the Unicorn's 2408 hard-disk recording system, which gives clear instructions and well-reproduced photos showing proper installation.

Alas, most such instructional material is not so good, and some provides so little information that it would be funny if it weren't so tragic. Many excel in some areas and fall short in others.

Producing good documentation costs manufacturers money, but doing so ultimately saves them money by cutting down on tech-support calls from users (if the users *read* the docs!). Furthermore, good documentation increases the popularity of a manufacturer's products by building its image as a Class Outfit.

Here are some ideas on documentation that I wish every manufacturer would consider implementing in order to provide truly useful information to customers.

1. Create owner's manuals that are comprehensive, well written, and well organized. This is basic, folks. Many manuals are incomprehensible or lack explanations of features. There is simply no excuse for this garbled stuff. Manuals should provide plenty of step-by-step procedures for essential operations; numerous graphics, including screen shots for software; a section that's organized by application (recording features, playback features, and so on); and a reference section that's organized by function (for example, by menu and command for software).

2. Compile a good index in the manual. A good index is the most important piece of documentation that a manufacturer can provide in a manual, and yet it is usually the lamest. Even if all else fails in terms of the manual's organization or writing, if it has a good index, you will at least be able to find the information that it contains. When I am in midsession and need to look up information about a feature, I turn to the index. Seven times out of ten it is of no use. Good indexes take a whole lot of work to create. Manufacturers: please, do the work.

3. Include a quick-reference guide. Because no one can memorize everything, having a quick-reference guide can be very helpful. Lexicon sends out a laminated card with its PCM 81/91 processors that shows how to move around in the basic modes. Digidesign, too, has a good quick-reference card for *Pro Tools*. Every program, digital processor, hard-disk recorder, or other device with some level of complexity needs to have something accompany it that users can grab in midsession to remind themselves about how to access a function.

4. Compose applications notes. The more you can show a customer how useful the product they've bought from you is, the more inclined that person will be to buy from you again. Applications notes are cheap to create, and they can be posted on a Web site. They can also be made available to retailers, who frequently need to read them themselves.

5. Put up a good Web site. A Web site doesn't have to be fancy; it just has to be done well. And a good site offers tremendous marketing potential. Some companies have lots of great background information, links to other sites, applications notes, and so forth. Maintaining a presence on e-mail user-discussion lists is another good way to get the word out about your product.

I'm aware that I'm asking manufacturers to do a lot of work. But I've been around this business for a while now, and in the Final Mix, good information is good product support, good product support is good marketing, and good marketing is just good business.

ILLUSTRATION BY STEVE CURL

Introducing the 308.

The newest member of the MOTU family.

If you need digital I/O for your computer, MOTU's new 308 delivers more than any other audio interface.

The 308 gives you three banks — 8 channels each — of optical S/PDIF, RCA S/PDIF and AES/EBU digital input and output.

And in today's recording studio, digital I/O means the cleanest possible link between your computer and all of the latest, hottest outboard gear. If you just bought a new pair of monitors, there's a good chance even they have digital inputs.

Best of all, you can mix and match 308, 1224 and 2408 I/Os (up to three at once) to meet your exact I/O needs — up to 72 simultaneous input and outputs!

So check out MOTU audio interfaces for the latest in digital and analog connectivity for your Mac or PC.

- The 308 is a 24-channel expansion I/O for MOTU 2408 and 1224 hard disk recording systems.
- 8 channels of optical S/PDIF digital I/O (24-bit).
- 8 channels of RCA S/PDIF digital I/O (24-bit).
- 8 channels of AES/EBU digital I/O (24-bit).
- Word Clock In and Out.
- Adds 24 channels of digital I/O to a core 2408 or 1224 hard disk recording system.
- Expandable — connect up to three 308 rack I/Os to a PCI-324 audio card for 72 inputs & outputs.
- ASIO and Wave drivers — compatible with all major Macintosh and Windows audio software.
- Suggested US retail price: $695.

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8 channels
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